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Reining

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(54) **MICROPHONE ARRANGEMENT HAVING MORE THAN ONE PRESSURE GRADIENT TRANSDUCER**

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(51) **Int. Cl.**
H04R 9/08 (2006.01)

(52) **U.S. Cl.**
USPC **381/92**; 381/122; 381/357

(58) **Field of Classification Search**
USPC 381/92, 122, 174, 357
See application file for complete search history.

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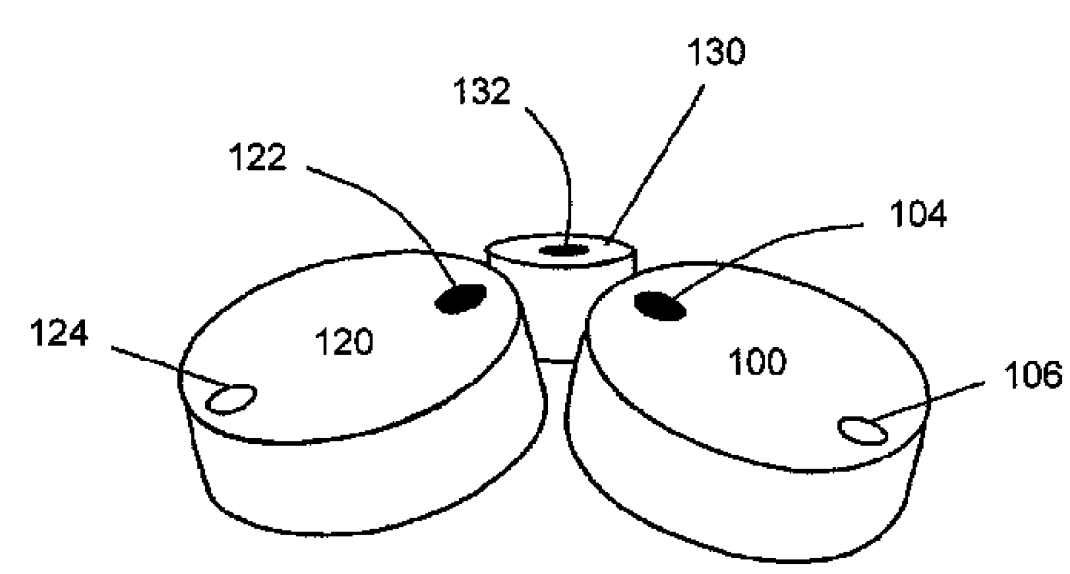
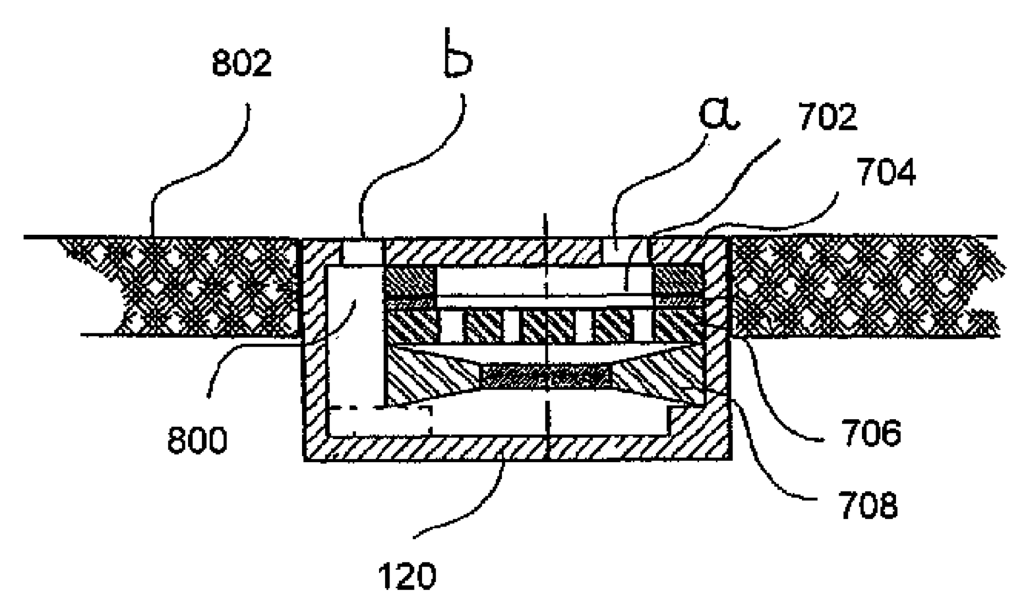
(57) **ABSTRACT**

A microphone arrangement includes multiple pressure gradient transducers having an acoustic center, a first sound inlet opening leading to a front of a diaphragm, and a second sound inlet opening leading the back of the diaphragm. A directional characteristic of the pressure gradient transducers includes an omni portion and a figure-eight portion. The pressure gradient transducers have a direction of maximum sensitivity in a main direction. Each main direction of the pressure gradient transducers is inclined. The acoustic center of a pressure transducer and the pressure gradient transducers are positioned within an imaginary sphere having a radius that corresponds to double the largest dimension of the diaphragm of one of the transducers.

15 Claims, 14 Drawing Sheets





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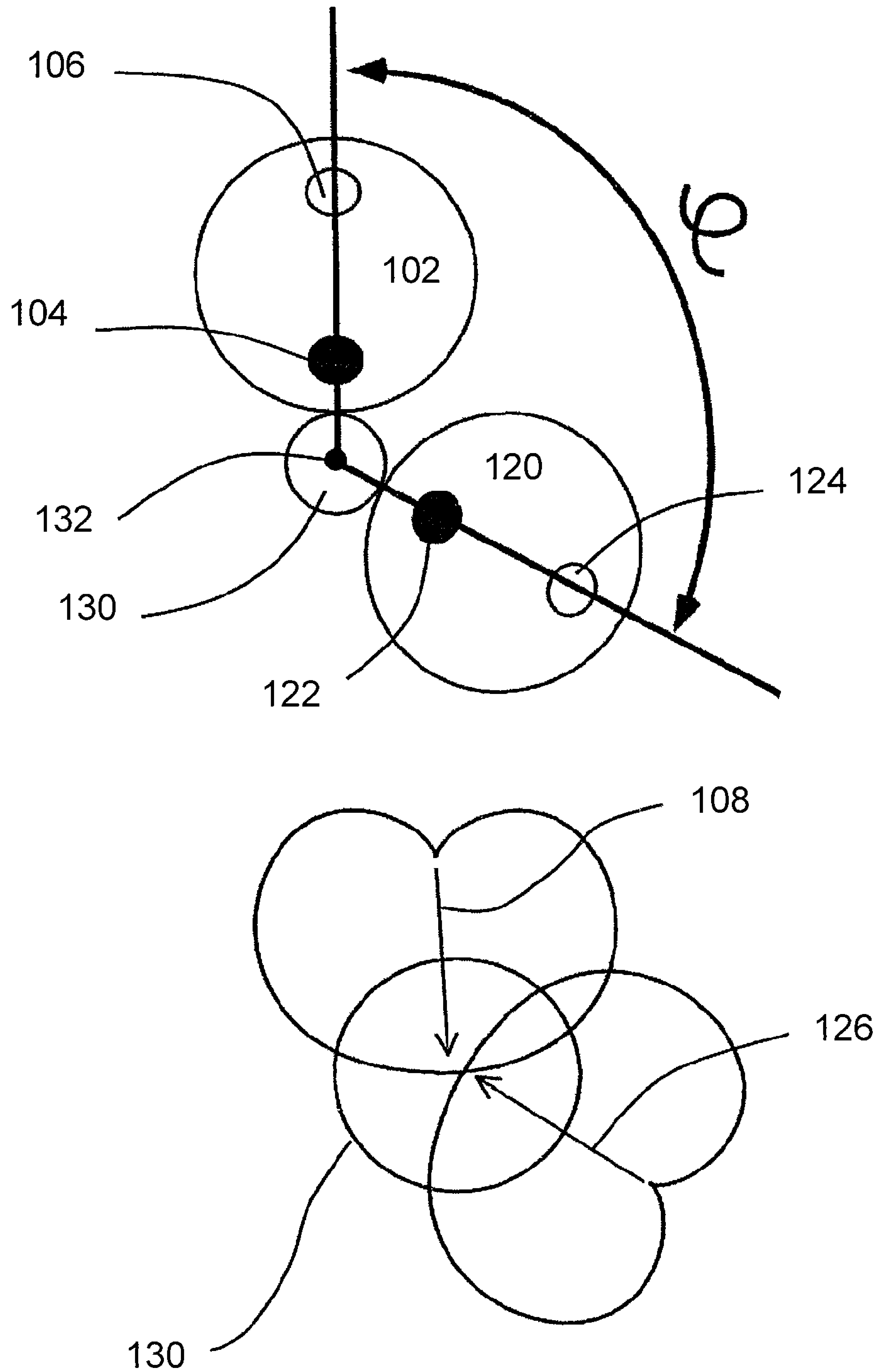

FIGURE 1

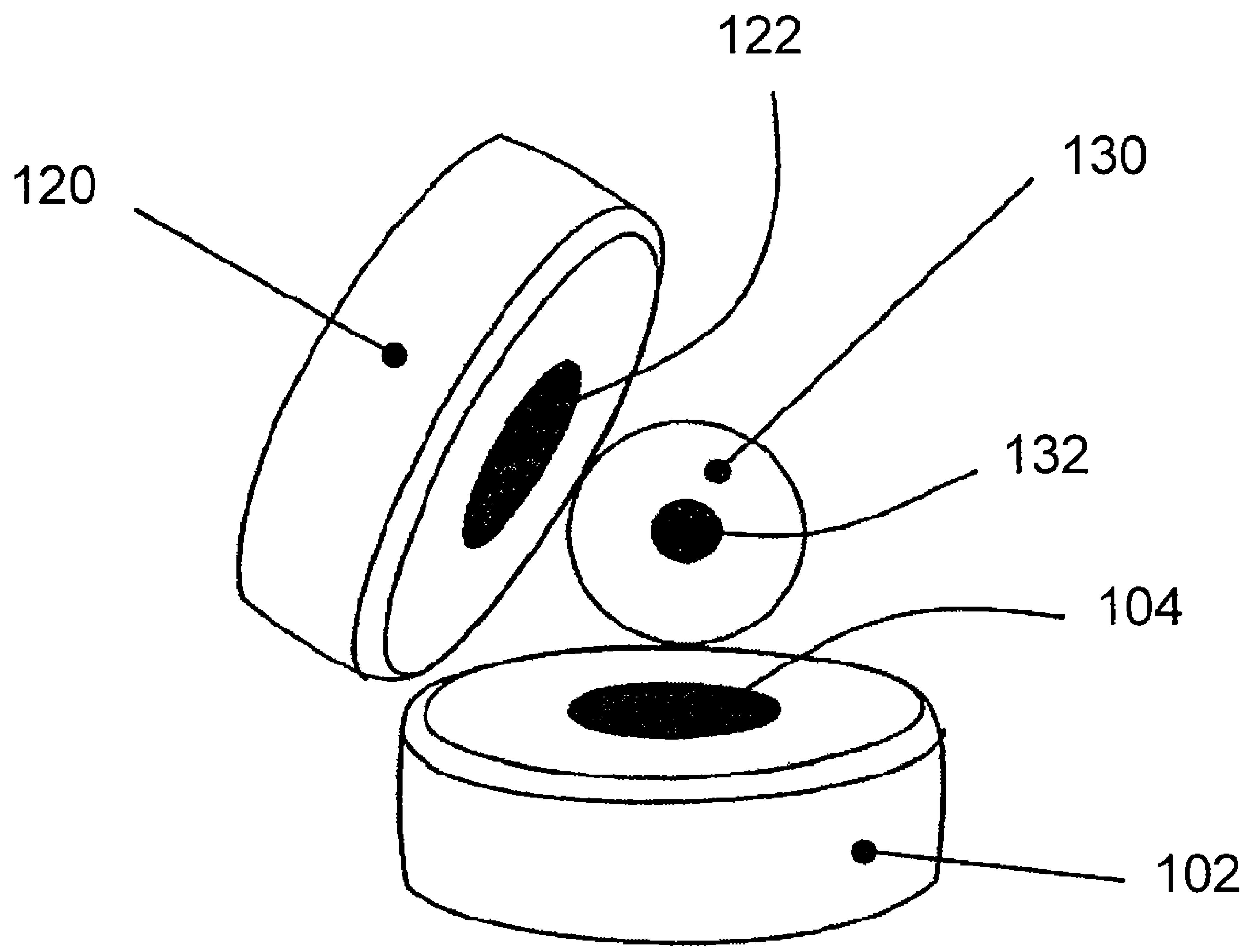

FIGURE 2

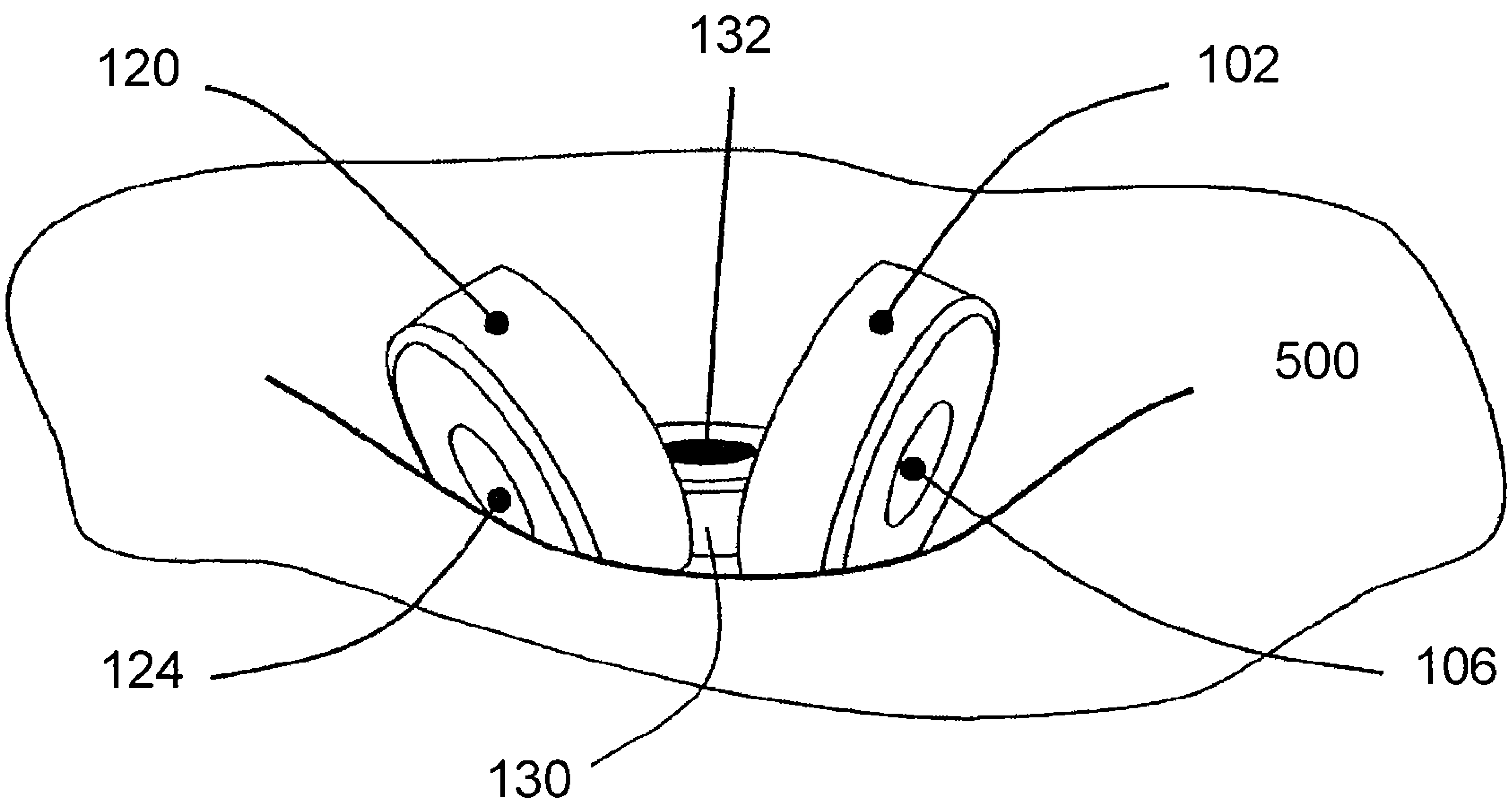

FIGURE 3

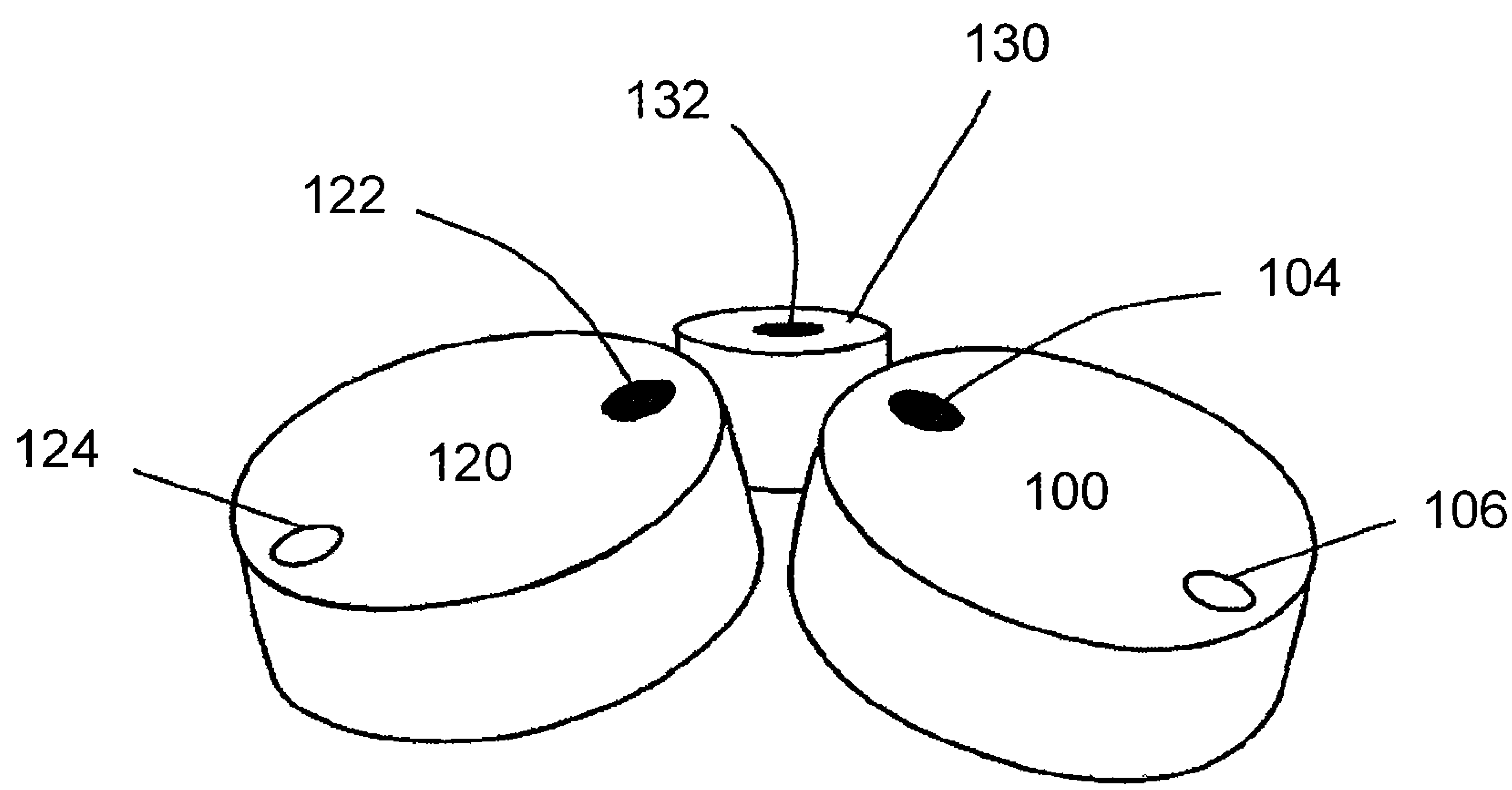

FIGURE 4

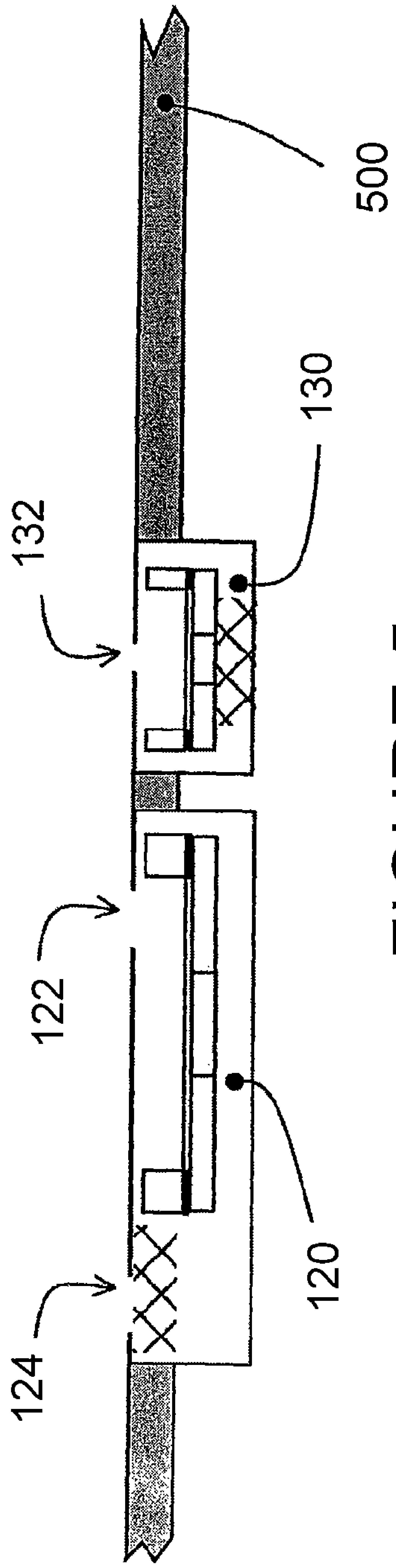

FIGURE 5

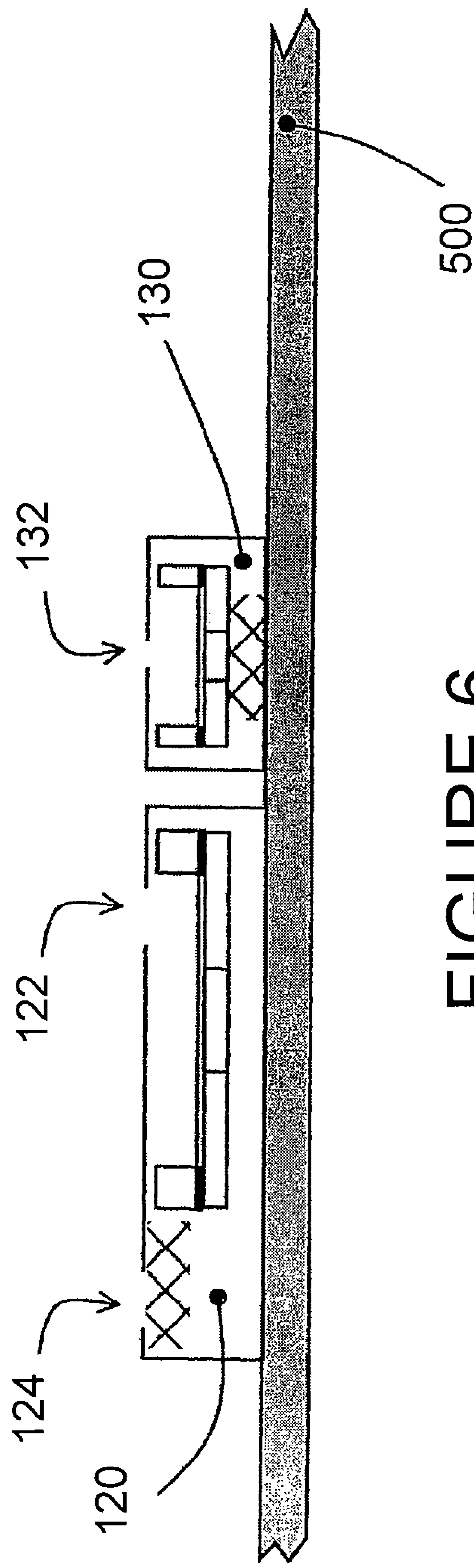

FIGURE 6

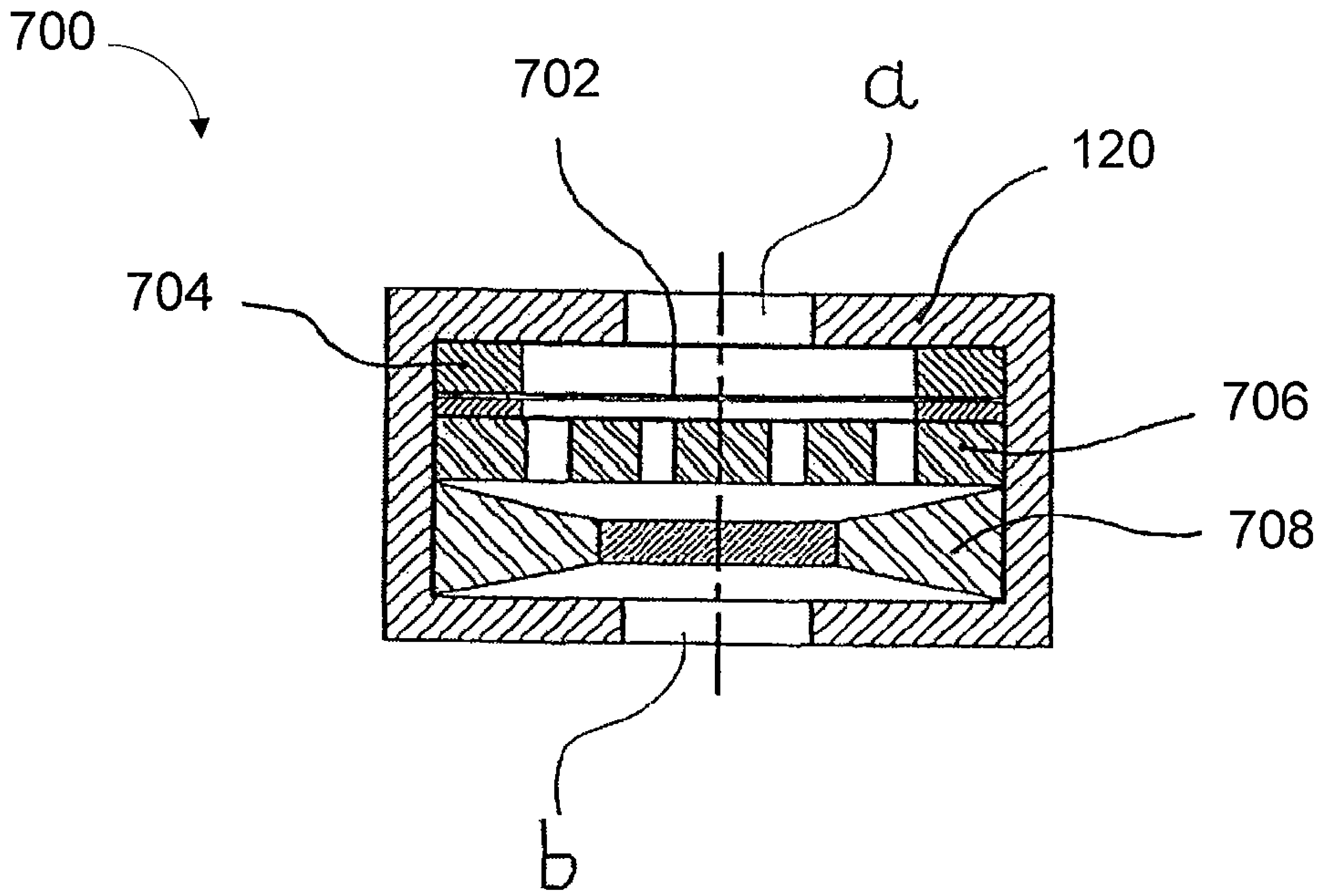

FIGURE 7

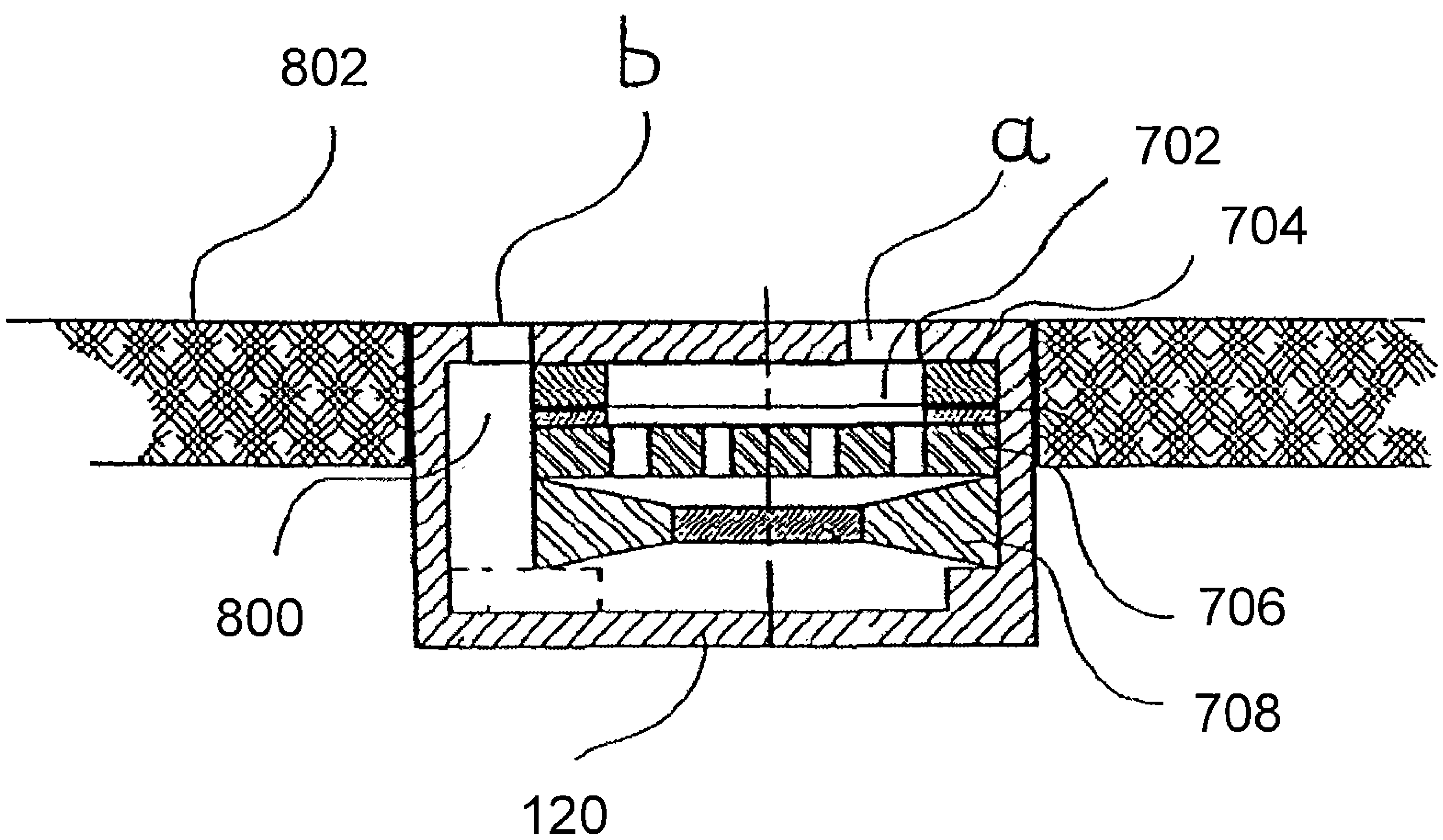

FIGURE 8

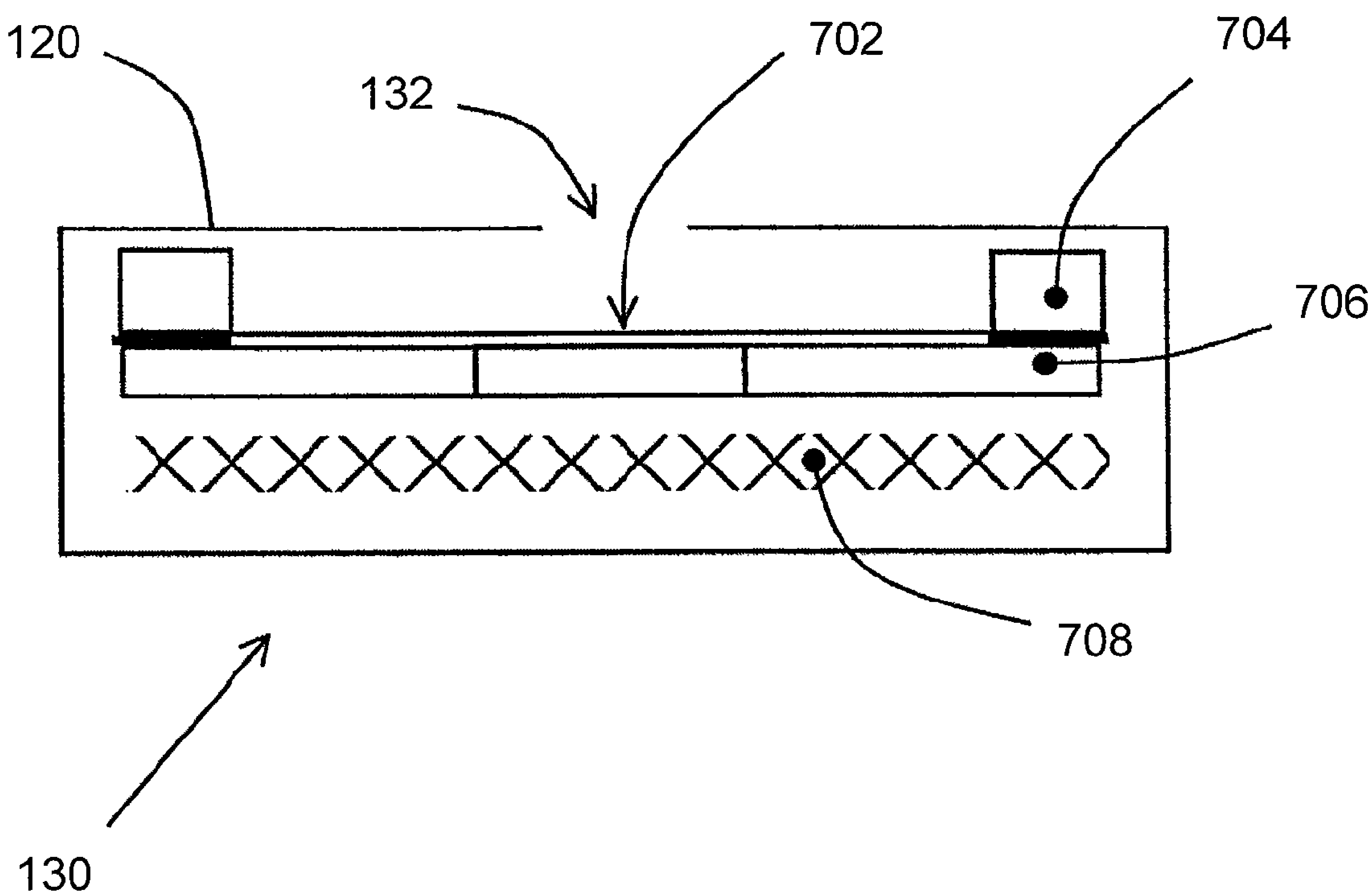

FIGURE 9

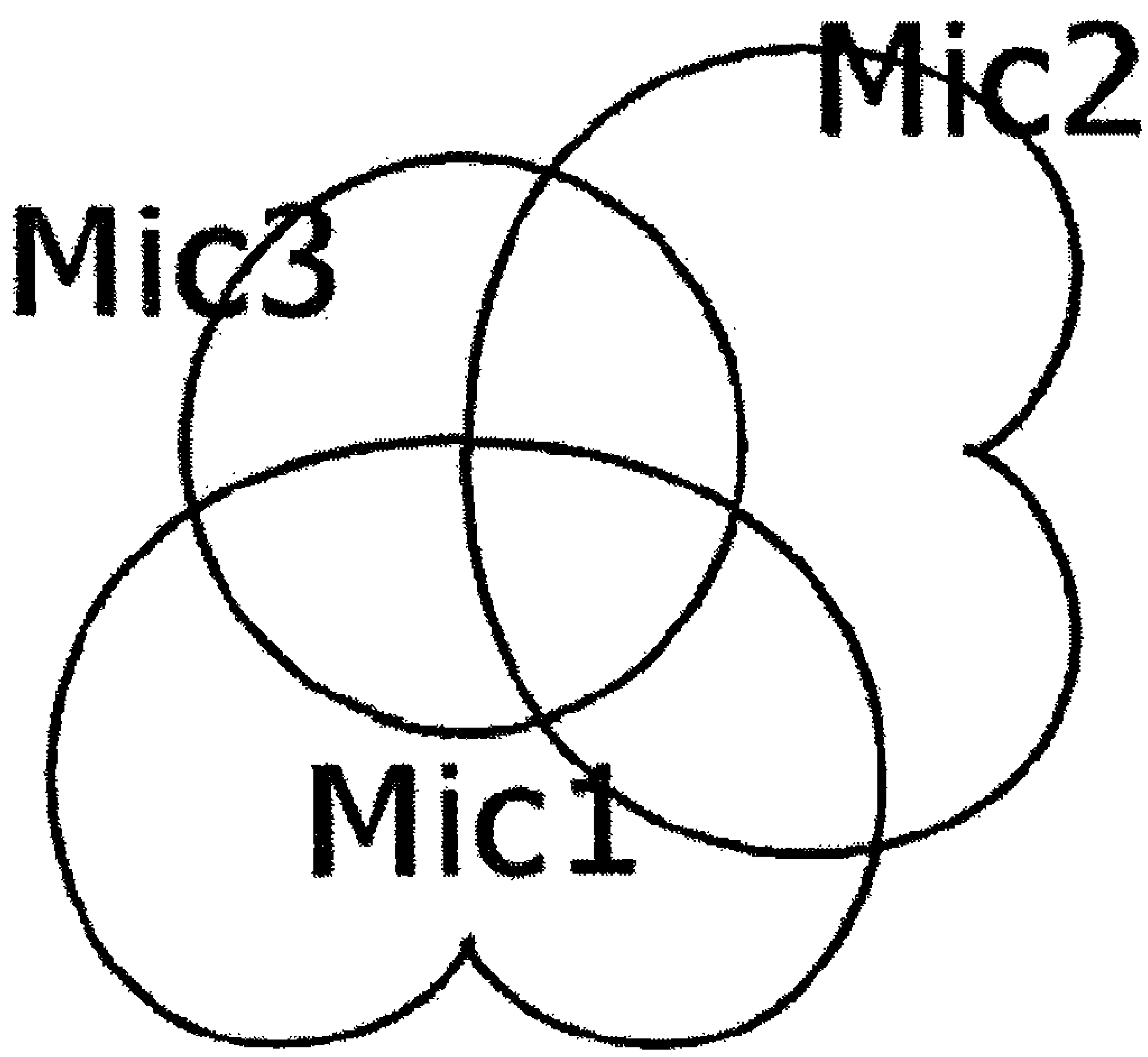

FIGURE 10

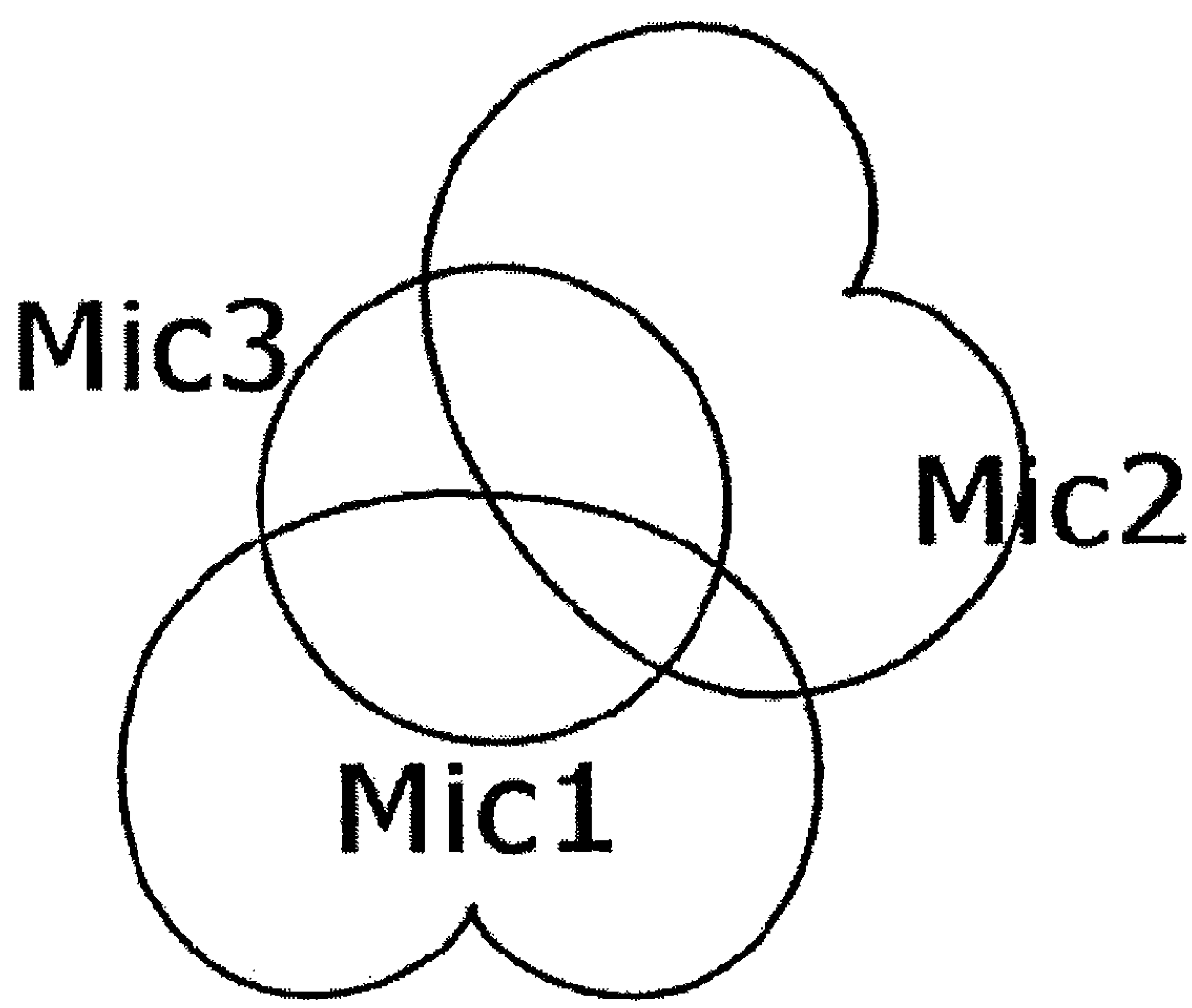

FIGURE 11

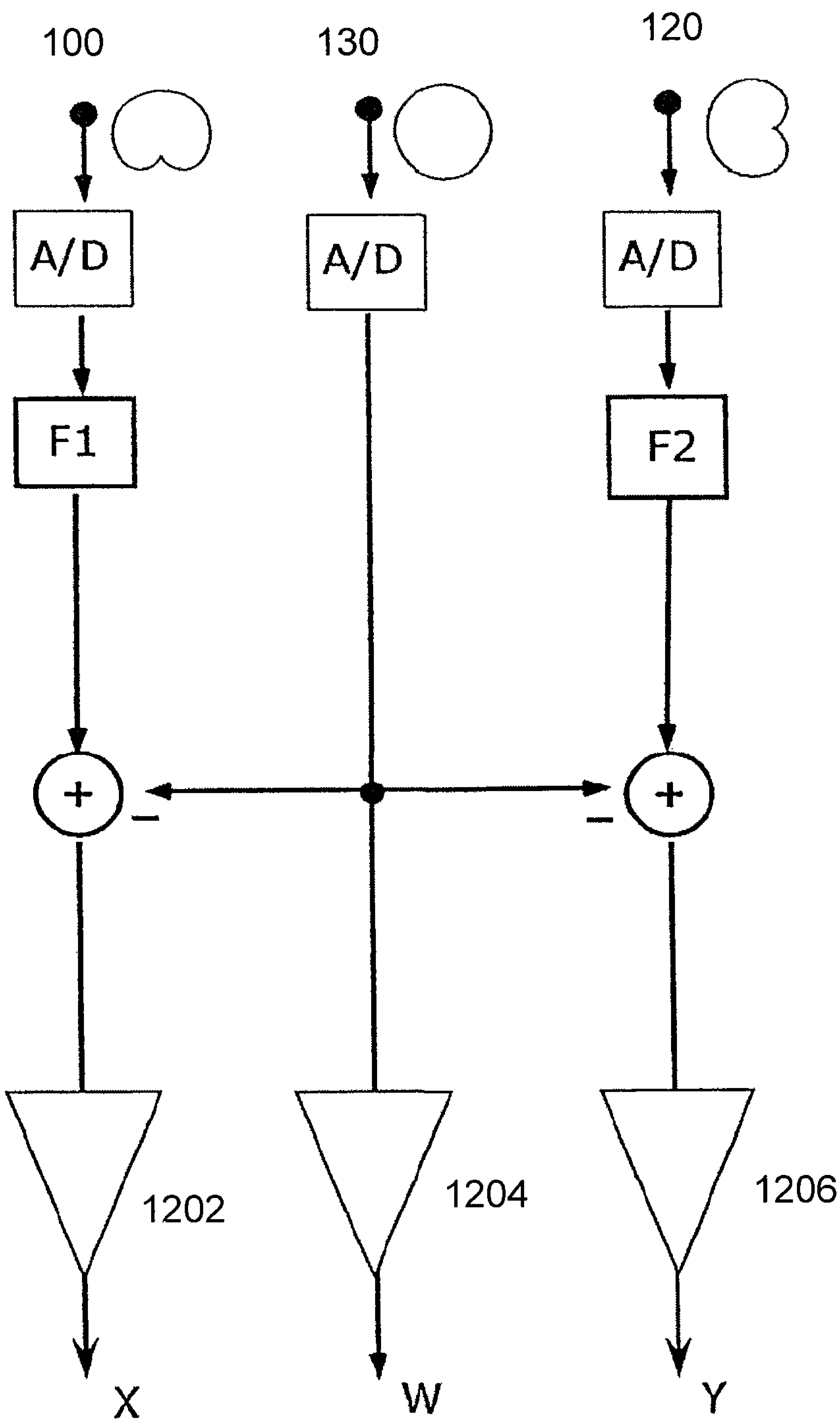

FIGURE 12

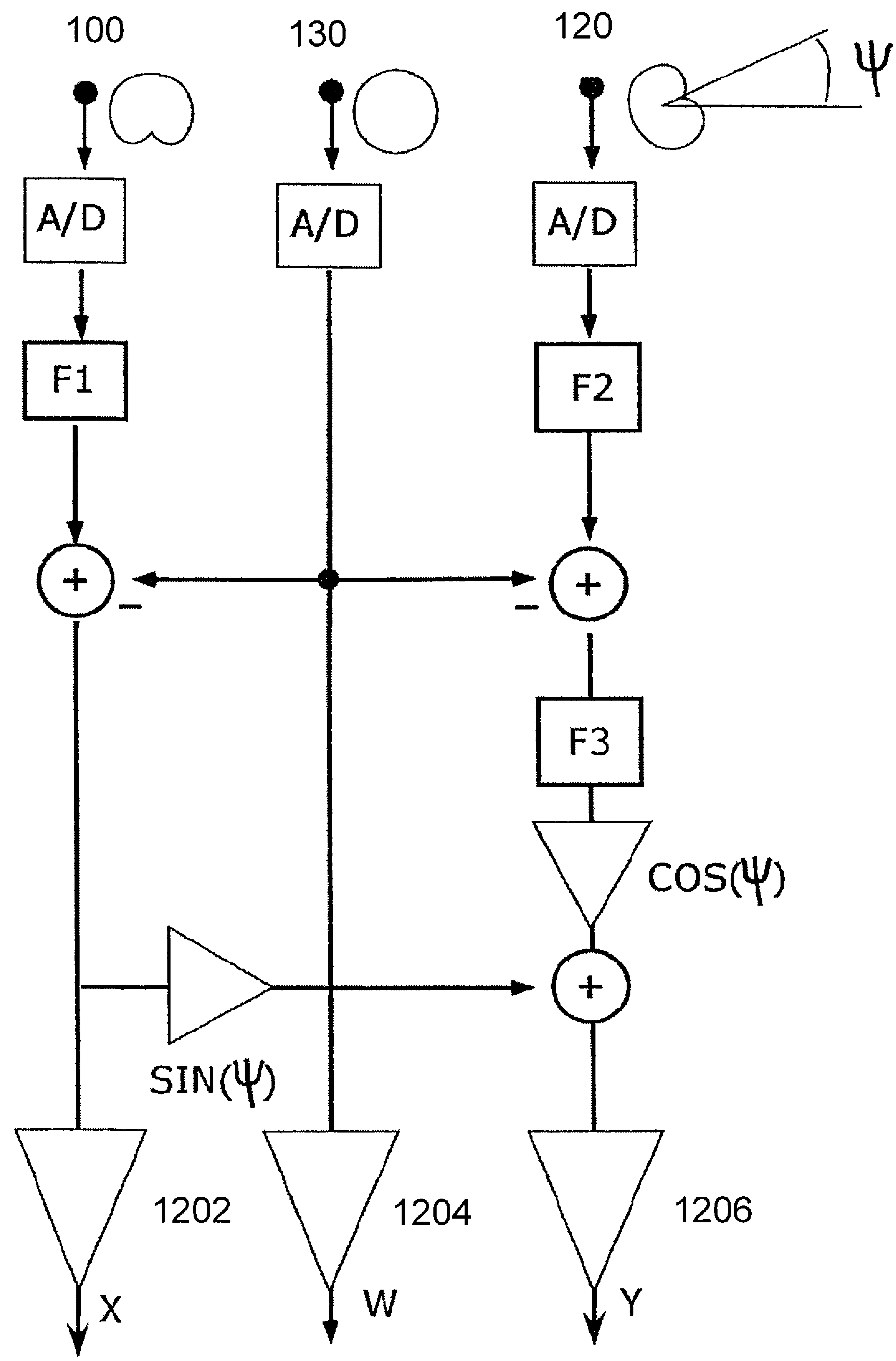

FIGURE 13

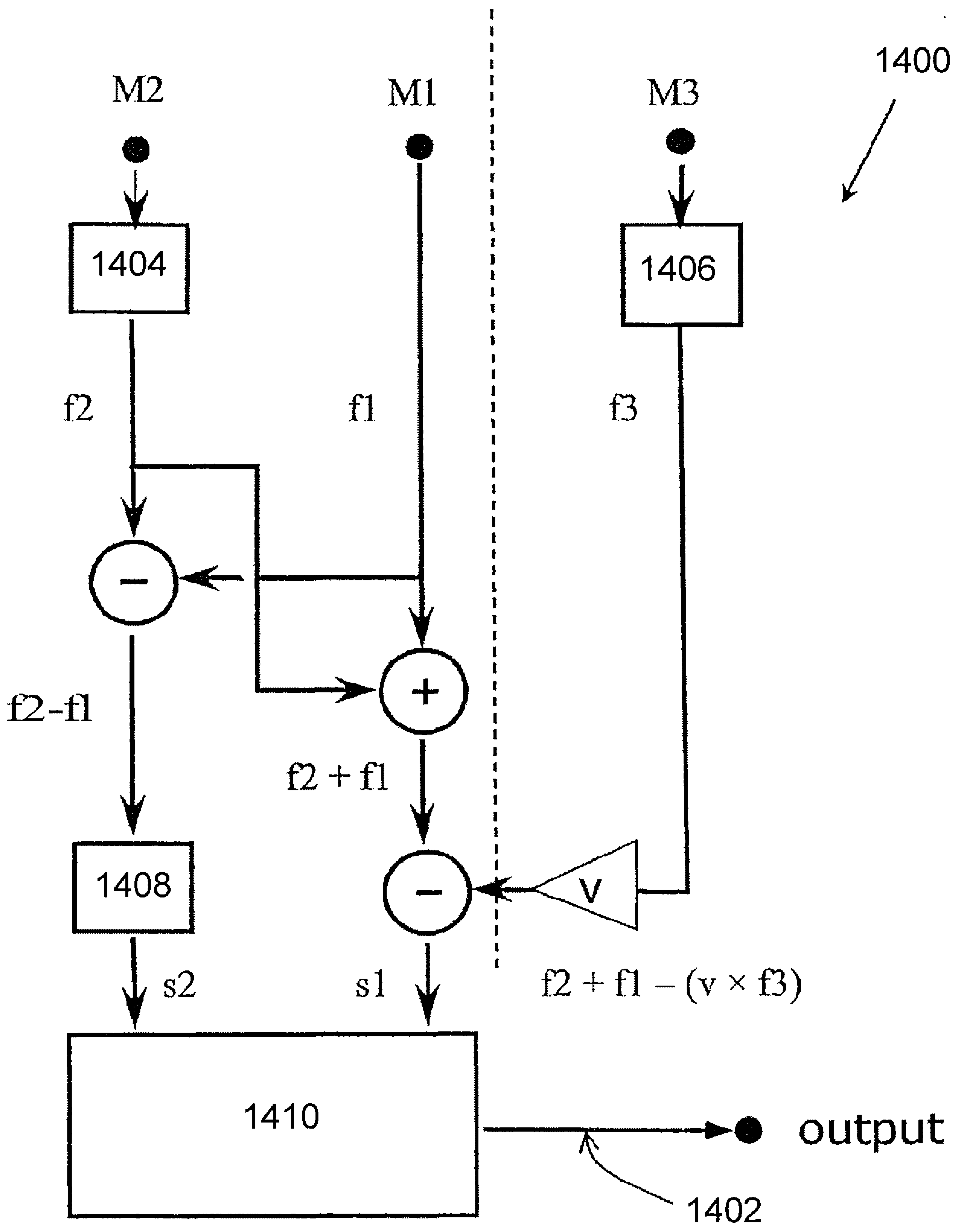
1400
M2
M1
M3
1404
1406
f2
f1
f3
f2-f1
f2 + f1
1408
V
s2
s1
f2 + f1 − (v × f3)
1410
output
1402

FIGURE 14

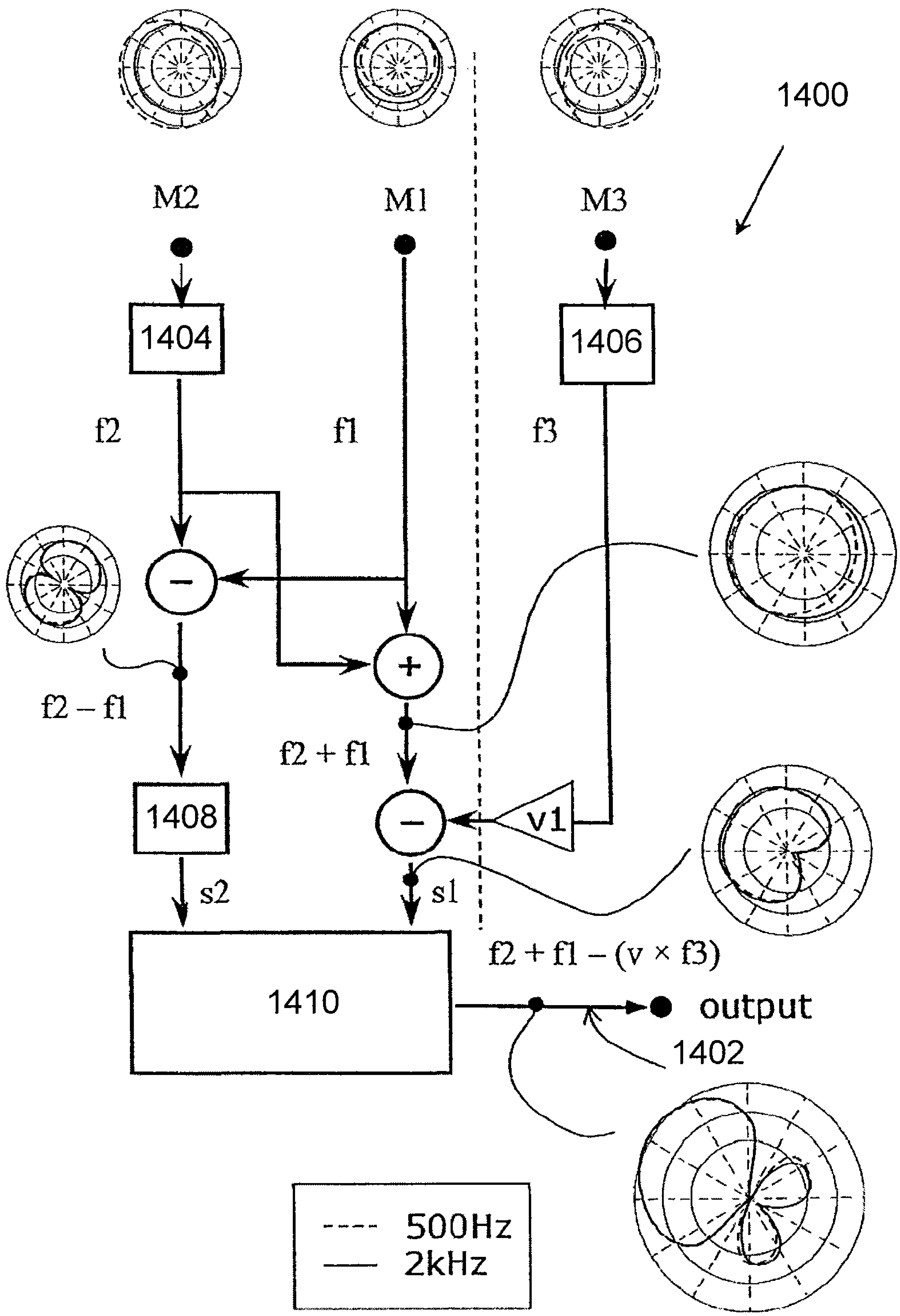
1400
M2
M1
M3
1404
1406
f2
f1
f3
f2 – f1
f2 + f1
1408
v1
s2
s1
1410
f2 + f1 – (v × f3)
output
1402
500Hz
2kHz

FIGURE 15

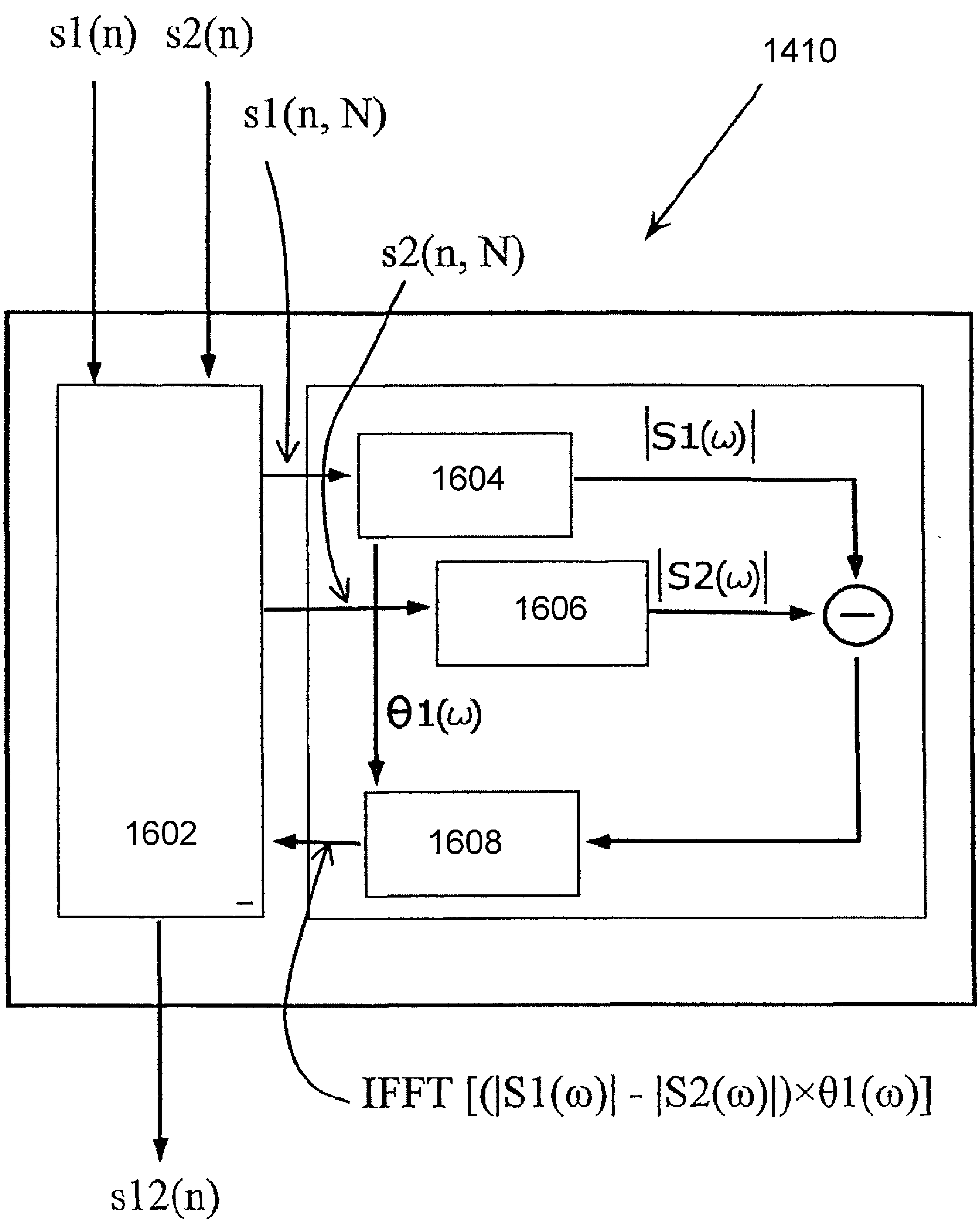
s1(n)
s2(n)
1410
s1(n, N)
s2(n, N)
|S1(ω)|
1604
|S2(ω)|
1606
θ1(ω)
1602
1608
IFFT [(|S1(ω)| - |S2(ω)|)×θ1(ω)]
s12(n)

FIGURE 16

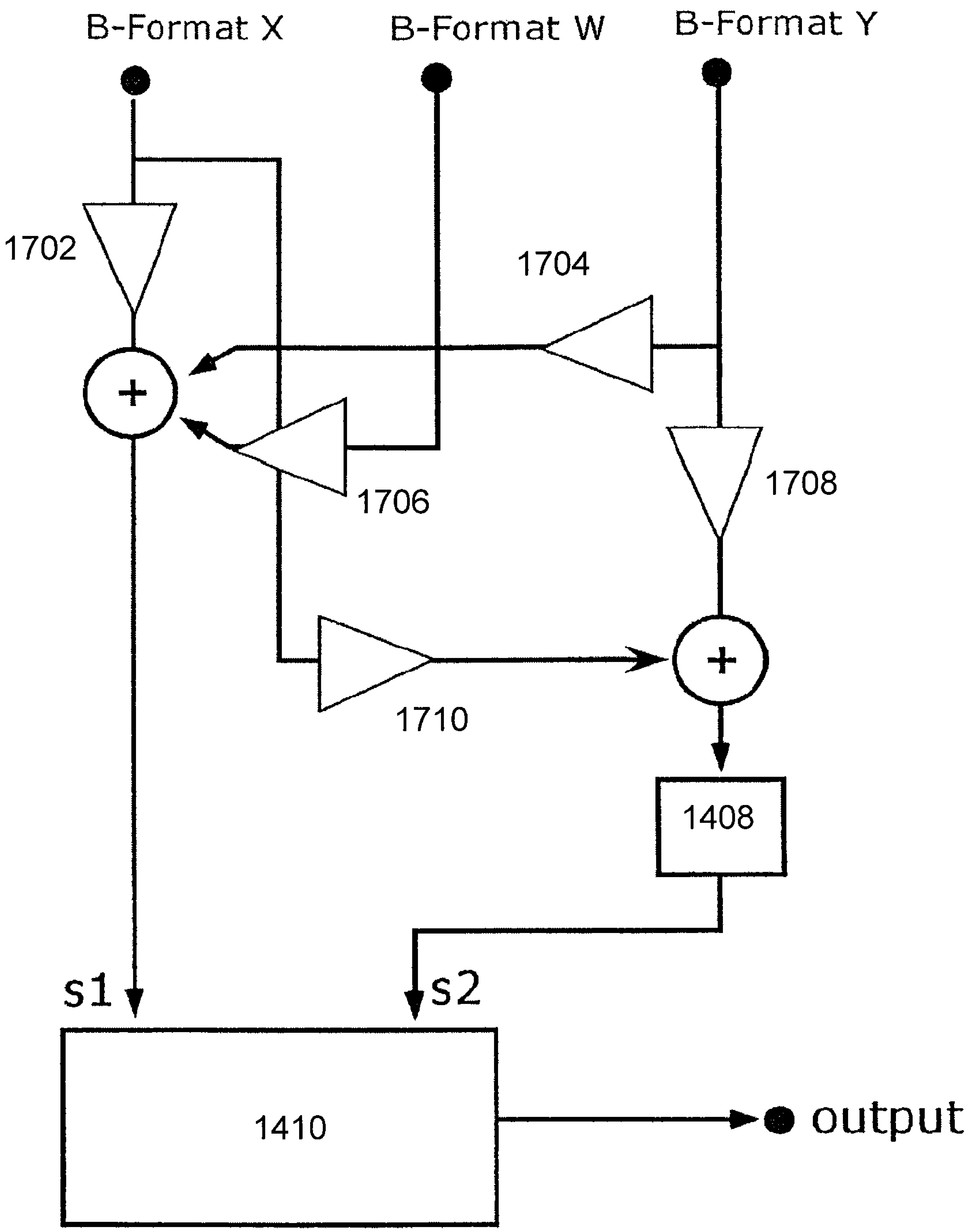
B-Format X
B-Format W
B-Format Y
1702
1704
+
1706
1708
+
1710
1408
s1
s2
1410
output

FIGURE 17

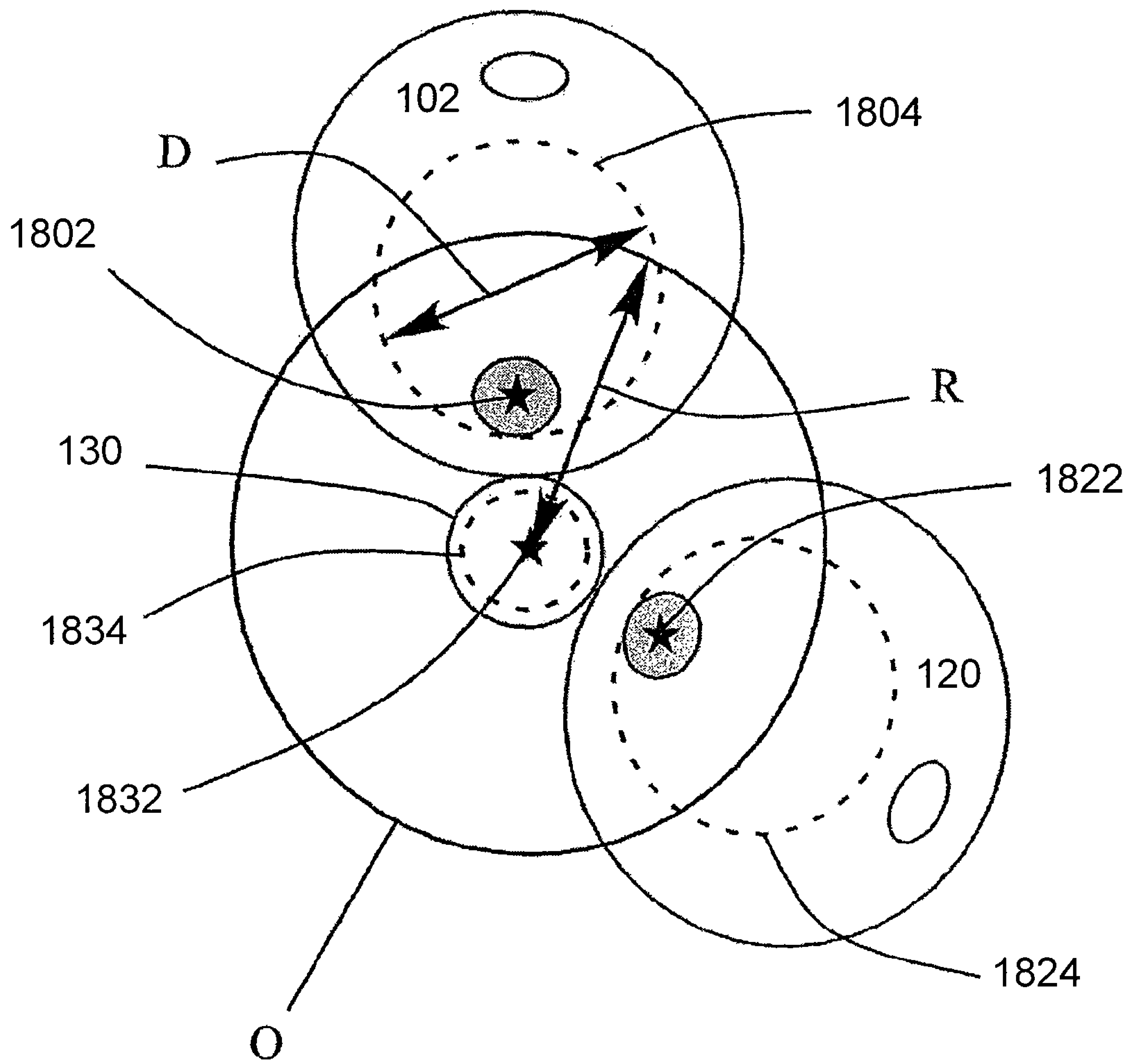

FIGURE 18

MICROPHONE ARRANGEMENT HAVING MORE THAN ONE PRESSURE GRADIENT TRANSDUCER

PRIORITY CLAIM

This application claims the benefit of priority from PCT/AT2007/000514, filed Nov. 13, 2007, which is incorporated by reference.

BACKGROUND OF THE INVENTION

1. Technical Field

This disclosure relates to devices that convert sound into operating signals and more specifically to microphones having multiple microphone arrangements.

2. Related Art

A coincident arrangement of gradient transducers may form a soundfield microphone (or B-format microphone). A soundfield microphone includes four pressure gradient capsules. The individual capsules are arranged in a tetrahedral form with the diaphragms of the individual capsules parallel to the imaginary surfaces of a tetrahedron. Each pressure gradient receivers delivers signals A, B, C or D and has a directional characteristic deviating from a sphere. The characteristic may be represented by $k+(1-k)\times\cos(\theta)$. θ denotes the azimuth, under which the capsule is exposed to sound. k indicates a percentage of omni signal (in a sphere, k=1, in a figure-of-eight, k=0). The signals of the individual capsules maybe denoted A, B, C and D. The axis of symmetry of directional characteristic of each individual microphone is perpendicular to the diaphragm and to the corresponding surface of the tetrahedron. The axes of symmetry of the directional characteristic of each individual capsule may form an angle of about 109.5° with each other.

According to one calculation, the four individual capsule signals maybe converted to a B-format (W, X, Y, Z):

$$W=\tfrac{1}{2}(A+B+C+D)$$

$$X=\tfrac{1}{2}(A+B-C-D)$$

$$Y=\tfrac{1}{2}(-A+B+C-D)$$

$$Z=\tfrac{1}{2}(-A+B-C+D)$$

The forming signals may form a sphere (W) and three figure-eights (X, Y, Z) that are orthogonal to each other. To configure the frequency and phase response of the directions, so that a flat energy characteristic is achieved with respect to the frequencies in the audible range, the signals W, X, Y, Z may be equalized. For a zero-order signal (W) and the first-order signals X, Y, Z, theoretical equalization characteristics depend on the frequency and effective distance of the center of the microphone capsules from the center of the tetrahedron.

The main directions of the figure eight X, Y, Z are normal to the sides of a cube enclosing the tetrahedron. By linear combination of at least two of these B-format signals, an arbitrary microphone capsule may be synthesized. Deviations from the theory based on the use of real capsules and the failure to satisfy ideally the coincidence requirement cause the performance of the synthesized microphones to deteriorate.

Synthesizing or modeling of the microphone may occur precisely in that the omni signal W is combined with one or more of the signals X, Y, Z, taking into account a linear weighting factor "r". For directional characteristics in the area between a sphere and a cardioid, this may be derived for a synthesized capsule in the X-direction through the formula $M=W+r\times X$, in which r can assume arbitrary values >0. The level of the signal M obtained may be normalized, so that the desired frequency trend is obtained for the main direction of the synthesized capsule. If a synthesized capsule is analyzed in any direction, additional weighting factors may be used, since rotation of the synthesized capsule in any direction may occur through a linear combination of three orthogonal figure-eights (X, Y, Z).

In a soundfield microphone the directional characteristic of the entire microphone may be adjusted. The microphone may be adapted even during playback or a final production of a sound carrier. It is possible to focus on a corresponding soloist of an ensemble, to mask out unexpected and undesired sound events by influencing the directional characteristic, or to follow a moving sound source (for example, a performer on the stage), so that the recording quality remains independent of the changed position of the sound source.

When sound is recorded from a soundfield microphone, the entire sound field may be described at any location in time. Time differences, etc., may be analyzed during selected evaluations. When deviations occur, the coincidence conditions for small wavelengths may no longer be satisfied. Distortions and artifacts may occur with respect to the frequency response and directional characteristic of a synthesized signal. A rotation of each individual gradient capsule of the soundfield microphone of about 180°, so that each of the four diaphragm surfaces is brought closer to the center, has shown that artifacts may not be eliminated at higher frequencies. Acoustic shadowing of the front microphone mouthpieces may not alter the limit frequency, up to which the calculation method applies.

There is a trade-off between the coincidence requirement and the attainable noise distance of the employed gradient capsules. The larger the individual diaphragm surface, the more noise distance may be achieved. However, this relationship leads to a larger distance of the diaphragm surfaces to the center of the arrangement. An optimal solution requires positioning of the four individual capsules as closely as possible to each other, so that the sound inlet on the back of the gradient transducer is influenced by the resulting structure of the closely positioned capsules. This means that the cavity formed in the interior of the microphone arrangement, and naturally also its delimitation by the microphone arrangement, as well as its mounts, etc., will act as an acoustic filter. The acoustic filter may affect the acoustic filtering by the sound paths that lead to the back of the individual capsules. The effect of this additional acoustic filter is frequency-dependent and may have its strongest effect at frequencies at which the wavelength of the sound is about the same order as the dimensions of the diaphragm or the dimensions of the entire soundfield microphone. In some soundfield microphones, this effect may occur in the frequency range around 10 kHz, at which rejection, (e.g., the frequency response from the direction from which the individual capsule is least sensitive becomes weakest and, drops below 10 dB).

In some soundfield microphones, two of the capsules may be situated with their main direction positioned downward, which means that they may be particularly sensitive to non-ideal microphone mounting or fastening under practical conditions. Such acoustic disturbances, based on the capsule arrangement, may develop due to reflections on the mounting material, on the floor, etc. In addition, the capsules in the close arrangement may be influenced when the theoretically rotationally symmetric directional characteristic of the synthesized omni signal is disturbed.

In some soundfield microphones a common configuration (X-Y-plane) is achieved by switching four capsule signals. The B-format signals in the X-Y-plane may be formed from microphone signals that meet at an angle of about 54° in each capsule under the influence of sound. If a directional diagram of a gradient transducer is considered, scattering of the rejection angle of the individual capsules may have a stronger effect, as the inlet direction deviates from the main direction. If two capsules with slightly different polar patterns exposed to sound from 0° differ only by the sensitivity described, at angles greater than 0°, the difference is increased by a percentage as a result of the difference in rejection angles.

SUMMARY

A microphone arrangement includes pressure gradient transducers having an acoustic center, a first sound inlet opening leading to a front of a diaphragm, and a second sound inlet opening leading the back of the diaphragm. A directional characteristic of the pressure gradient transducers includes an omni portion and a figure-eight portion. The pressure gradient transducers have a direction of maximum sensitivity in a main direction. Each main direction of the pressure gradient transducers is inclined. The acoustic center of a pressure transducer and the pressure gradient transducers are positioned within an imaginary sphere having radius that corresponds to about double the largest dimension of the diaphragm of one of the transducers.

Other systems, methods, features, and advantages will be, or will become, apparent to one with skill in the art upon examination of the following figures and detailed description. It is intended that all such additional systems, methods, features and advantages be included within this description, be within the scope of the invention, and be protected by the following claims.

BRIEF DESCRIPTION OF THE DRAWINGS

The system may be better understood with reference to the following drawings and description. The components in the figures are not necessarily to scale, emphasis instead being placed upon illustrating the principles of the invention. Moreover, in the figures, like referenced numerals designate corresponding parts throughout the different views.

FIG. **1** is a microphone arrangement.

FIG. **2** is an alternative microphone arrangement.

FIG. **3** is an alternative microphone arrangement in which the transducers of FIG. **2** are embedded within a boundary.

FIG. **4** is a microphone arrangement with gradient transducers and a pressure transducer arranged on an omni surface.

FIG. **5** shows a cross-section through the microphone arrangement of FIG. **1**.

FIG. **6** shows transducers embedded in the boundary.

FIG. **7** is a gradient transducer with sound inlet openings on opposite sides of a capsule housing.

FIG. **8** is a gradient transducer with sound inlet openings on the same side of the capsule housing.

FIG. **9** is a pressure transducer in cross-section.

FIG. **10** is the directional characteristics of three transducers, in which the main directions of the pressure gradient transducers enclose an angle of about 90°.

FIG. **11** shows the directional characteristics of three transducers, in which the main directions of the pressure gradient transducers enclose an angle of about 120°.

FIG. **12** is a block diagram that renders the B-format signals from the arrangement of FIG. **10**.

FIG. **13** is a block diagram that renders the B-format signals from an arrangement of FIG. **11**.

FIG. **14** is an expanded signal processor.

FIG. **15** shows the directional characteristics of FIG. **14**.

FIG. **16** is a spectral subtraction.

FIG. **17** is a signal processing controller.

FIG. **18** shows coincidence.

DETAILED DESCRIPTION OF THE PREFERRED EMBODIMENTS

A microphone arrangement converts individual transducers signals into a B-format. Coincidence may be ensured or improved. Shadowing effects that may arise when the individual capsules shadow may be reduced or substantially attenuated. Acoustic disturbances from spatial conditions near the microphone arrangement and the dependence on capsule tolerances (for example, deviations in the manufacturing process) may be minimized.

In some microphone arrangements the acoustic centers of the pressure gradient capsules and the pressure transducer lie within an imaginary sphere whose radius corresponds to about double the largest dimension of the diaphragm of a gradient and/or pressure transducer. The criterion ensures the coincident position of the transducers. The acoustic centers of the pressure gradient transducers and the pressure transducer lie within an imaginary sphere whose radius corresponds to the largest dimension of the diaphragm of a transducer. Coincidence may increase by moving the sound inlet openings together.

“Synthesized directional characteristic” is a combination of individual B-format signals, for example, a sphere (W) with at least one additional B-format signal (a figure-eight), and also their further processing, such as equalization, bundling, rotation, etc. The individual signals may be processed by a weighting of the signals.

The expression “directional characteristic” is not only the directional characteristic of the real capsules, but of signals in general. These signals may be comprised of other signals (for example, B-format signals) and have complicated multiple directional characteristics. If such directional characteristics may not be achieved with individual capsules, the expression “directional characteristic” is used to establish from which spatial areas the formed or synthesized signals preferably yield acoustic information.

FIG. **1** is a microphone arrangement comprising two pressure gradient transducers **102** and **120** and a pressure transducer **130**. The directional characteristic of the pressure gradient transducer may include an omni portion and a figure-eight portion. This directional characteristic may be represented as $P(\theta)=k+(1-k)\times\cos(\theta)$, in which k denotes the angle-independent omni portion and $(1-k)\times\cos(\theta)$ denotes the angle-dependent figure eight-portion. In FIG. **1**, the microphone arrangement may include a gradient transducer that renders a cardioid characteristic. Alternative arrangements may generate other characteristics. The characteristics may be derived from the combination of a sphere and figure-eight, for example, (e.g., hypercardioids, supercardoid, shotgun, etc).

The directional characteristic of the pressure transducer **130** may comprise an omni characteristic. Deviations from an omni shape may occur at higher frequencies due to variances in manufacturing tolerances and quality. In these arrangements directional characteristics may be described as a sphere or sphere like. A pressure transducer, in contrast to the gradient transducer, has only one sound inlet opening. The deflection of the diaphragm is proportional to pressure.

Deflection may not occur due to a pressure gradient, between the front and back of the diaphragm.

The gradient transducers **102** and **120** may lie in a two dimensional plane (e.g., an x-y-plane). The main directions **108**, **126** (the directions of maximum sensitivity) are inclined relative to each other by the azimuthal angle ϕ (lower portion of FIG. **1**). The angle ϕ between two main directions preferably takes on values between about 30° and about 150°. A preferred angle lies at about 90°. At 90°, two signals are orthogonal to each other. In alternative systems any type of gradient transducer may be used including a flat transducer or boundary microphone, in which the two sound inlet directions lie on the same side surface, e.g., the boundary.

FIG. **7** and FIG. **8** show the difference between a "normal" gradient capsule and a "flat" gradient capsule. In FIG. **7**, a sound inlet opening "a" is situated on the front of the capsule housing **700** and a second sound inlet opening "b" is situated on the opposite back side of the capsule housing **700**. The front sound inlet opening "a" is connected to the front of diaphragm **702**, which is tightened through a diaphragm ring **704**, and the back sound inlet opening "b" is connected to the back of diaphragm **702**. In the pressure gradient capsules the front of the diaphragm may receive sound with little resistance (e.g., relatively unhampered). The back of the diaphragm of the pressure gradient may only be reached through the passage of an acoustically phase-rotating element of the sound. In many microphone arrangements, the sound path to the front is shorter than the sound path to the back. The arrows show the path of the sound waves to the front or back of the diaphragm **702** in FIG. **9**. In the area behind electrode **706**, there may be an acoustic friction element **8**. The acoustic friction element may form of a constriction. It may comprise a non-woven material or a foam, for example.

In the flat gradient capsule of FIG. **8**, (e.g., a boundary microphone) sound inlet openings a, b are positioned on the front of the capsule housing **700**, in which one leads to the front of diaphragm **702** and the other leads to the back of diaphragm **700** through a sound channel **800**. The transducer may be incorporated in a boundary **802**, like, a console of a vehicle. The acoustic friction element **708** may comprise, for example, a non-woven element, a foam material, constrictions, perforated plates, etc., that may be arranged in the area next to diaphragm **702**. The acoustic friction element **708** may take a very flat form.

By arranging inlet openings a, b on one side of the capsule, a directional characteristic asymmetric to the diaphragm axis may be generated, (e.g. cardioid, hypercardioid, etc.). The capsules may include characteristics described in EP 1 351 549 A2 and the corresponding U.S. Pat. No. 6,885,751 A, which are incorporated by reference.

A pressure transducer, or a zero-order transducer, is shown in FIG. **9**. In this arrangement, the front of the diaphragm **702** is connected to the surroundings in zero-order transducers. The back faces a closed volume. The pressure transducer has an essentially omni directional characteristic. Slight deviations are obtained as a function of the frequency.

In FIG. **1**, two gradient capsules **102**, **120** are oriented toward each other. The sound inlet openings **104** and **122** leading to the front of the corresponding diaphragm lie adjacent or very near each other and the sound inlet openings **106**, **124** leading to the back of the diaphragm lie on the periphery of the arrangement. The point of intersection of the lengthened connection lines that join the front sound inlet opening **104** and **122** to the rear sound inlet opening **106** and **124** is viewed as the center of the microphone arrangement. In the lower portion of FIG. **1**, this is the center, toward which the main directions **108** and **126** are directed. The front sound inlet openings **104**, **122** of the two transducers **102** and **120**, or mouthpieces are situated in the center area of the arrangement. The coincidence of the two transducers may be strongly influenced by this arrangement. The pressure transducer **130** is now situated in the center area of the microphone arrangement, in which the single-sound inlet opening of the pressure transducer **130** is preferably situated at the intersection of the connection lines of the sound inlet openings of pressure gradient transducers **102**, **120**.

Coincidence may occur due to the acoustic centers of the gradient transducers **102**, **120** and the pressure transducer **130** that may be positioned together. In some arrangements, the center may occur at a common or same point. The acoustic center of a reciprocal transducer may be the point from which omni waves seem to be diverging when the transducer is acting as a sound source. "A note on the concept of acoustic center", by Jacobsen, Finn; Barrera Figueroa, Salvador; Rasmussen, Knud; Acoustical Society of America Journal, Volume 115, Issue 4, pp. 1468-1473 (2004), which is incorporated by references, examines various ways of determining the acoustic center of a source, including methods based on deviations from the inverse distance law and methods based on the phase response. "The acoustic center of laboratory standard microphones" by Salvador Barrera-Figureueroa and Knud Rasmussen; The Journal of the Acoustical Society of America, Volume 120, Issue 5, pp. 2668-2675 (2006), which is incorporated by reference also describes how acoustic centers may be identified.

The acoustic center may be determined by measuring spherical wave fronts during sinusoidal excitation of the acoustic transducer. The measurement may occur at a selected frequency in a certain direction and at a certain distance from the transducer in a small spatial area. The area may be an observation point. Analyses of the spherical wave fronts may identify the center of the omni wave—the acoustic center.

For a reciprocal transducer, such as a condenser microphone, the transducer may be operated as a sound emitter or sound receiver. The acoustic center may be identified by:

$$p(r) = j\frac{\rho * f}{2 * r_t} M_f * i * e^{-\gamma * r_i} \tag{1}$$

r_i ... Acoustic center

ρ ... Density of air

f ... Frequency

M_f ... Microphone sensitivity

i ... Current

γ ... Complex wave propagation coefficient

In pressure receivers exclusively, the center may comprise average frequencies (in the range of 1 kHz), that may deviate at high frequencies. The acoustic center may occur in a small area. The acoustic center of gradient transducers may be identified by a different approach, since formula (1) does not consider the near-field-specific dependences. The location of an acoustic center may also be identified by locating the point in which a transducer must be rotated to observe the same phase of the wave front at the observation point.

In a gradient transducer an accurate center may be identified through a rotational symmetry. The acoustic center may be situated on a line normal to the diaphragm plane. The center point on the line may be determined by two measurements, at a point most favorably from the main direction of about 0°, and at a point of about 180°. In addition to the phase responses of these two measurements, for an average estimate of the acoustic center in the time range used, one method alters the rotation point around which the transducer is rotated between measurements. The adjustment may ensure that the impulse responses are maximally congruent (e.g., so that the maximum correlation between the two impulse responses lies in the center).

In some microphone arrangements, in which the two sound inlet openings are situated on a boundary, the acoustic center may not be the diaphragm center. The acoustic center may lie closest to the sound inlet opening that leads to the front of the diaphragm. This forms the shortest connection between the boundary and the diaphragm. In other arrangements, the acoustic center could also lie outside of the capsule.

When using an additional pressure transducer, separation may be considered. A prerequisite for separation of the omni signal portion from the figure-eight signal portion of a pressure gradient capsule, through an external pressure transducer, in addition to coincidence, is also the constancy of the omni characteristic as a gauge of the quality of the obtainable separation of the omni signal portion and the figure-eight signal portion.

If one considers the diaphragm of a pressure transducer lying in the XY-plane, and refers to the angle that any direction in the XY-plane encloses with the X-axis is the azimuth, and refers to the angle that any direction encloses with the XY-plane as the elevation, the following may be practiced. The deviation of the pressure transducer signal from an ideal omni signal may be greater with an increasing frequency (for example, above 1 kHz), but increases much more strongly during sound exposure from different elevations. Based on these considerations, an alternative is obtained when the pressure transducer is arranged on a boundary, so that the diaphragm is substantially parallel to the boundary. In another alternative, the diaphragm lies as close as possible to the boundary, preferably aligned with it, but at least within a distance that corresponds to the maximum dimension of the diaphragm. Such alternatives yield a particularly high degree of separation quality of the omni signal portion and the figure-eight signal portion. The acoustic center for such incorporation lies in a line substantially normal to the diaphragm surface at or near the center of the diaphragm. The acoustic center, with good approximation, may lie on the diaphragm surface at the center.

The coincidence criterion requires, that the acoustic centers **1802**, **1822**, **1832** of the pressure gradient transducers **102**, **120** and the pressure transducer **130** lie within an imaginary sphere O, whose radius R is double of the largest dimension D of the diaphragm of a transducer.

In some alternative microphone arrangements the acoustic centers of the pressure gradient transducers and the pressure transducer may lie within an imaginary sphere having radius corresponding to the largest dimension of the diaphragm of a transducer. By increasing the coincidence through movement of the sound inlet openings together, exceptional results may be achieved.

To ensure a coincidence condition, the acoustic centers **1802**, **1822**, **1832** of the pressure gradient capsules **102** and **120** and the pressure transducer **130** lie within an imaginary sphere O, having a radius R equal to the largest dimension D of the diaphragm of a transducer. The size and position of the diaphragms **1804**, **1824**, **1834** are indicated in FIG. **18** by dashed lines.

In an alternative, this coincidence condition may also be established in that the first sound inlet openings **104**, **122** and the sound inlet opening for the pressure transducer **130** lie within an imaginary sphere whose radius corresponds to the largest dimension of the diaphragm of a transducer. Since the size of the diaphragm determines the noise distance and therefore represents a direct criterion for designing the acoustic geometry. The largest diaphragm dimension (for example, the diameter in a round diaphragm, or a side length in a triangular or rectangular diaphragm) may determine the coincidence condition. In other systems the diaphragms **1804**, **1824**, **1834** do not have the same dimensions. In these systems, the largest diaphragm is used to determine the preferred criterion.

In the microphone arrangement of FIG. **1**, the two gradient transducers **102**, **120** are arranged in a plane. The connection lines of the individual transducers, which connect the front and rear sound inlet openings to each other, are inclined with respect to each other by an angle of about 120°.

FIG. **4** is an alternative microphone arrangement, in which the two pressure gradient transducers **100**, **120** and the pressure transducer **130** are not arranged in a plane, but on an imaginary omni surface. This may occur when the sound inlet openings of the microphone arrangement are arranged on a curved boundary, for example, like the console of a vehicle. The boundary, in which the transducers are embedded, or on which they are fastened, is not shown in FIG. **4**.

In the curvature arrangement of FIG. **4**, the distance to the center is reduced (which is desirable, because the acoustic centers lie closer together), but the mouthpiece openings may be somewhat or partially shaded. A curved arrangement may alter the directional characteristic of the individual capsules to the extent that figure-eight portion of the signal becomes smaller (from a hypercardioid, then a cardioid). To minimize the adverse effect of shadowing, the curvature may be limited (e.g., not exceed 60°). The pressure gradient capsules **100**, **120** may be positioned on the outer surface of an imaginary cone whose surface line encloses with the cone axis an angle of at least 30°.

The sound inlet openings **104**, **122** that lead to the front of the diaphragm and the sound inlet opening **132** of the pressure transducer may lie in a plane, referred to as base plane. The sound inlet openings **106**, **124**, in an arrangement on a curved boundary, lie outside of this base plane. The projections of the main directions of the two gradient transducers **100**, **120** into the base plane enclose an angle that is preferably between about 30° and about 150°, with a preferred angle of about 90°.

Like the arrangement where the capsules are arranged in the plane, the main directions of the pressure gradient transducers are inclined relative to each other by an azimuth angle ϕ, (e.g., they are not only inclined relative to each other in a plane of the cone axis, but the projections of the main directions are also inclined relative to each other in a plane normal to the cone axis).

In the arrangement of FIG. **4**, the acoustic centers of the two gradient transducers **100**, **120** and the pressure transducer **130** also lie within an imaginary sphere whose radius corresponds to double (or about double) the maximum dimension of the diaphragm of the transducer. By this spatial proximity of the acoustic centers, the coincidence that may be the B-format, is achieved. As in the alternatives to FIG. **1**, the capsules depicted in FIG. **4** are also arranged on a boundary, or embedded within it.

Boundary arrangements of capsules are shown in FIGS. **5** and **6**. The capsules in FIG. **5**, which shows a section through the microphone arrangement from FIG. **1**, sit on the boundary **500** or are fastened to it, whereas in FIG. **6**, they are embedded in boundary **500** and are flush (or substantially flush) with their fronts with boundary **500**.

In another alternative system, the pressure gradient capsules **100**, **120** and the pressure transducer **130** are arranged within a common housing, with the diaphragms, electrodes, and mounts of the individual transducers being separated from each other by intermediate walls. The sound inlet openings may no longer be seen from the outside. The surface of the common housing, in which the sound inlet openings are arranged, may be a plane (referring to an arrangement of FIG. **1**) or a curved surface (referring to an arrangement according to FIG. **2**). The boundary may be a plate, console, wall, cladding, etc.

FIGS. **2** and **3** show another alternative arrangement that is constructed without a one-side sound inlet microphone. In each of the pressure gradient transducers **102**, **120**, the first sound inlet openings **104**, **122** are arranged on the front of the capsule housing and the second sound inlet openings **106**, **124** are arranged on the back of the capsule housing. The pressure transducer **130** has only the sound inlet opening **132** on the front. The first sound inlet openings **104**, **122**, which lead to the front of the diaphragm, face each other and again. They lie within an imaginary sphere whose radius is equal to about double the largest dimension of the diaphragm of the pressure gradient transducer. The main directions (shown as arrows in FIG. **1**) of the two gradient transducers have the microphone arrangement in a common center area. The projections of the main directions, in a plane in which the first sound inlet openings **104**, **122** and their center and the sound inlet opening **132** of the pressure transducer **132** lie, established as the base plane, enclose an angle of about 30° to 150°, but preferably 90° or about 90°, with each other. Deviations of ±10° lie within the scope of these arrangements.

FIG. **3** shows an alternative in which the gradient capsules from FIG. **2** are embedded within a boundary **500**. It must then be kept in mind that the sound inlet openings are not covered by the boundary **500**.

In signal flow, the partial signals W, X, Y, applied in the most often used B-format may be formed from only three capsule signals. A set of signals, including an omni signal and at least two figure eight signals, may now be viewed in a generalized manner as the B-format.

Initially, the formation of a B-format occurs in a microphone arrangement in which the main directions **108**, **126** of the two gradient transducers **102**, **120** or their projections, in a base plane, enclose an angle of about 90° with each other in the case of a curved boundary. The directional characteristics of the individual transducers **102**, **120**, **130** in such an arrangement are shown in FIG. **10**.

FIG. **12** shows how a B-format is formed from the individual capsule signals K**1**, K**2** and K**3**. The B-format includes an omni signal W, an X-component of the B-format, and a Y-component of the B-format. The system extracts a figure-eight signal from each of the gradient signals of transducers **102**, **120**. This occurs in that the omni portion of the gradient signal is taken off from the gradient signal through the omni signal of the pressure transducer.

The calculation starts from linear frequency responses. W, X, and Y may be described as:

$$W=K3$$

$$X=K1-K3$$

$$Y=K2-K3$$

and may be implemented through software stored on a local or remote computer readable storage medium (executed by a processor or signal processor) or the circuit shown FIG. **13**. W is the omni signal and X and Y is the orthogonal figure-eight signals.

To subtract the entire omni portion of the gradient signal, normalization of the individual transducer signals may be required. This normalization may occur in different ways, for example, as described.

The characteristics of each individual gradient capsule may also be described by

$$Kx = \frac{1}{a_x + b_x}(a_x + b_x\cos(\theta)) \qquad (1)$$

in which a_x represents the weighting factor of the omni portion and b_x represents the weighting factor for the gradient portion. For values $a_x=1$, $b_x=1$, cardioid is obtained; for values $a_x=1$ and $b_x=3$, we obtain a hypercardioid is obtained.

If such normalization is carried out, the B-format signals assume the shape described by:

$$W = K3$$

$$X = K1 - K3 * \frac{a_1}{a_1 + b_1}$$

$$Y = K2 - K3 * \frac{a_2}{a_2 + b_2}$$

The directional characteristic of the employed gradient capsules is included in these formulas.

In some applications, gradient capsules may not always be used as a point of departure. They have a linear frequency response over the entire frequency range, and, have a frequency response that differs only in the level during sound exposure from another direction. In these applications, the signals are filtered. In these applications, the signals are filtered before the signals are further processed as shown in FIG. **12**. A correction factor F**1** is initially calculated, so that the gradient transducer **100**, during sound exposure from the main direction (direction of the maximum sensitivity) of the gradient transducer **120**, yields the same signal as pressure transducer **3**.

The filter coefficients and filter F**2** are calculated in the same way. The gradient transducer signal K**2** is adapted, so that it may yield the same signal as pressure transducer **130** when exposed to sound from the main direction of the gradient transducer **100** and vice-versa.

Any level differences and/or frequency response differences may then be carried out in channels X, Y and W by the calculation of a corresponding filter **1202**, **1204**, **1206** (according to the downward-directed triangles).

Formation of the B-format is described below when the main directions of the two gradient microphones **100**, **200** have an angle different from about 90° relative to each other. This is further explained by the example shown in FIG. **11** having an angle of about 120°.

If the participating gradient transducers are not situated at an angle of about 90° relative to each other, an additional calculation may ensure that ultimately two substantially orthogonal figure-eight characteristics remain.

FIG. **13** shows how this expansion may occur. The signals are initially adjusted by filtering of the microphone signals of the two gradient transducers **102**, **120** with filters F**1** and F**2**, so that after subtraction of the pressure transducer signal from the gradient transducer signal, only the figure-eight portion is left. Another filter F**3** ensures that the frequency response of the two figure-eights is identical in the main direction. From each figure-eight signal R_ψ, which was extracted from the signal of the gradient transducer **120** by subtraction of the omni portion, and whose directional characteristic is rotated by angle ψ from the 90° position of FIG. **14** a figure-eight signal Y should now be formed, which is orthogonal to the figure-eight signal X of the radiant transducer **102**. This occurs by a weighting and superposition of the two figure-eights X and R_ψ. The B-format signals W, X, Y so obtained are:

$$W = K3$$

$$X = K1 - K3 * \frac{a_1}{a_1 + b_1}$$

$$Y = X \times \sin(\psi) + R_\psi \times \cos(\psi),$$

in which

$$R_\psi = K2 - K3 * \frac{a_2}{a_2 + b_2}$$

Any level differences and/or frequency response differences could be carried out in channels X', Y' and W' by calculation with the corresponding filters **1202**, **1204**, **1206** (downward-facing triangles).

W represents the omni signal, which is an essentially omnidirectional signal. X and Y each represent a figure-eight lobe, whose axis of symmetry is parallel to the plane of the microphone. X and Y are substantially orthogonal to each other and are therefore inclined by about 90° relative to each other. By the combination of omni signal W with at least one of the figure-eight signals X, Y, any arbitrary directional characteristic may be generated. By a linear combination of X and Y with corresponding weighting factors, the figure-eight may be rotated within the x-y-plane. By a linear combination of this rotated figure-eight with the omni signal, the main direction of the synthesized signal may be rotated in different directions.

This linear combination may be written as the synthesized signal

$$M(q,r,s)=q\times W+r\times X+s\times Y,$$

in which q, r, s represent the weighting factors, with which the B-format signals are incorporated in the final signal M.

Synthesized microphone signals M**1**, M**2**, and optionally M**3**, serve as the basis, which are calculated by

$$M(q,r,s)=q\times W+r\times X+s\times Y.$$

The synthesized signals M**1**, M**2** and M**3** now have directional characteristics. These are cardioids, whose main directions lie in one plane and are inclined to each other by about 120°.

FIG. **14** shows a schematic block diagram between outputs, at which the synthesized signals M**1** and M**2** lie, and shows the output **1402** of the signal processing unit **1400**. The synthesized signals are initially digitized with A/D converters (not shown). Subsequently, the frequency responses of the synthesized signals are compared to each other, to compensate for manufacturing tolerances of the individual capsules. This occurs by linear filters **1404**, **1406**, which adjust the frequency responses of the synthesized signals M**2** and M**3** to those of synthesized signal M**1**. The filter coefficients of linear filters **1404**, **1406** are determined from the impulse responses of all participating gradient transducers, with the impulse responses being measured from an angle of about 0°, from the main direction. An impulse response is the output signal of a transducer, when it is exposed to a narrowly limited acoustic pulse in time. In determining the filter coefficients, the impulse responses of transducers **102** and **130** are compared with that of transducer **102**. The result of linear filtering of FIG. **14** is that the impulse responses of all gradient transducers **102**, **120**, **130**, after passing through the filter, have the same frequency response. This arrangement serves to compensate for deviations in the properties of the individual capsules.

Subsequently in the block diagram, a sum signal f**1**+f**2** and a difference signal f**1**−f**2** are formed from the filter signals f**1** and f**2** that result from M**1** and M**2** by filtering. The sum signal is dependent on the directional characteristic and its orientation in space and therefore dependent on the angle of the main directions of the individual signals M**1**, M**2** relative to each other, and contains a more or less large omni portion.

At least one of the two signals f**1**+f**2** or f**2**−f**1** is now processed in another linear filter **1408**. This filtering serves to adjust these two signals to each other, so that the subtraction signal f**2**−f**1** and the sum signal f**1**+f**2**, which has an omni portion, have the maximum possible agreement when overlapped.

In this analysis, the subtraction signal f**2**−f**1**, which has a figure-eight directional characteristic, is inflated or contracted in filter **1408** as a function of the frequency, so that maximum rejection in the resulting signal occurs when it is subtracted from the sum signal. The adjustment in filter **1408** occurs for each frequency and each frequency range separately.

Determination of the filter coefficients of filters **1408** also occurs via the impulse responses of the individual transducers. Filtering of the subtraction signal f**2**−f**1** gives the signal s**2**; the (optionally filtered) summation signal f**1**+f**2**—in the practical example with only two synthesized signals M**1**, M**2**—gives the signal s**1** (the portion of the signal processing unit **30**, shown on the right side of the dashed separation line, is not present during use of only two signals M**1**, M**2**).

However, three synthesized signals M**1**, M**2**, M**3** may be evolved in signal processing (to the right of the separation line in FIG. **14**). The signal f**3**, adjusted to the frequency response of signal M**1** in linear filter **1406**, is now multiplied by amplification factor v and subtracted as v×f**3** from the sum signal f**1**+f**2**. The resulting signal s**1** now corresponds to (f**1**+f**2**)−(v×f**3**), in the case of three signals.

It may be initially established by the application factor v, in which the useful direction should lie, (e.g., that spatial direction that is to be strongly limited by the directional characteristic of the synthesized total signal). The possible useful directions are unrestricted, in principle, because the synthesis signals M**1**, M**2** and M**3** may be arbitrarily rotated. For example, if factor v is very small, the effect of the third synthesis signal M**3** on the total signal is limited and the sum signal f**1**+f**2** dominates relative to signal v×f**3**. If, on the other hand, the amplification factor v is negative and large, the individual signal v×f**3** dominates over the sum signal f**1**+f**2** and the useful sound direction, or the direction in which the synthesized total signal directs its sensitivity. Therefore, that signal rotated by 180° with reference to the former case. By variation of factor v, this system permits a change in sum signals, so that an arbitrary directional characteristic is generated in the desired direction.

This bundling mechanism may be applied to all signal combinations. For the direction to which bundling is supposed to occur, an intrinsic spectral subtraction block may be applied. The signal processing steps occurring before the spectral subtraction block may be combined to the extent that only factor v need be different for two opposite directions. The other preceding steps and branches remain the same for these two directions.

The spectral subtraction applied to the two intermediate signals **s1** and **s2** occurs in block **1410**. FIG. **16** shows the individual components of a spectral subtraction block **1410** in detail in a digital domain. The A/D conversion of the signals also can only occur before spectral subtraction block **1410** and the filtering and signal combinations conducted before occurs on the analog plane.

Two signals **s1**(n) and **s2**(n) serve as the input of block **1410** in the time range derived from the signals that were recorded at the same time and at the same point (or at least in the immediate vicinity). This guarantees the coincident arrangement of transducers **102**, **120**, **130**. **s1**(n) represents the signal that has the most useful signal portions, whereas **s2**(n) represents the signal that contains more interference signals, in which signal **s2**(n) is characterized by the fact that it has a zero position, in the viewing of the polar diagram, in the useful sound direction; and n represents the sample index, and s(n) therefore corresponds to a signal in the desired time range.

The unit marked **1602** generates individual blocks with a block length N=L+(M−1) from the continuously arriving samples. L represents the number of new data samples in the corresponding block. The remainder (M−1) of samples was also already found in the preceding block. This method is "overlap and save" method.

The N samples contained in a block are then conveyed to unit designated **1604** at the times at which M−1 samples have reached unit **1602** since the preceding block. Unit **1604** is characterized by processing that occurs in a block-oriented manner. The signal **s1**(n, N) packed into blocks reaches unit **1604**, the unit **1606** is provided for the signal **s2**(n, N) packed into blocks in a similar way.

In units **1604**, **1606**, the end samples of signals **s1** and **s2** combined into a block, are transformed by FFT (fast Fourier transformation), for example, DFT (discrete Fourier transformation), into the frequency range. The signals **S1**(ω) and **S2**(ω) that form are broken into value and phase, so that the value signals |**S1**(ω)| and |**S2**(ω)| occur at the output of units **1604** and **1606**. By spectral subtraction, the two value signals are now subtracted from each other and produce (|**S1**(ω)|−|**S2**(ω)|).

Subsequently, it applies that the resulting signal (|**S1**(ω)|−|**S2**(ω)|) is transformed back to the time domain. For this purpose, the phase **Θ1**(ω), which was separated in unit **1604** from signal **S1**(ω)=|**S1**(ω)|×**Θ1**(ω) and which, like the value signal |**S1**(ω)|, also has a length of N samples, is used during the back-transformation. The back-transformation occurs in the one unit **1608** through an IFFT (inverse fast Fourier transformation), for example, IDFT (inverse discrete Fourier transformation) and is carried out based on the phase signal **Θ1**(ω) of **S1**(ω). The output signal of unit **53** can therefore be represented as IFFT [(|**S1**(ω)|−|**S2**(ω)|)×exp(**Θ1**(ω)].

The generated N samples of long digital time signal **S12**(n, N) is fed back to processing unit **1602**, where it is incorporated in the output data stream **S12**(n) according to the calculation procedure of the overlap and save method.

The parameters obtained in this method are block length N and rate (M−1)/fs [s] (with sampling frequency fs), with which the calculation or generation of a new block is initiated. In principle, in any individual sample, an entire calculation may be carried out, provided that the calculation unit is fast enough to carry out the entire calculation between two samples. In some conditions, about 50 ms has proven useful as the value for the block length and about 200 Hz as the repetition rate, in which the generation of a new block is initiated.

The signal processing just described (FIGS. **14** and **15**), in which a signal narrowly bundled in a specific direction can be produced, starting from B-format signals, may also be implemented more simply and directly. FIG. **17** shows a corresponding circuit for three B-format signals W, X, Y to the synthesized signals **s1** and **s2**. The subsequent spectral subtraction block **1410** remains. The amplifiers **1702** to **1710** weigh the individual B-format signals according to the direction in which one intends to direct a narrow lobe of the directional characteristic. The filter **1408** ensures that during the spectral subtraction of signal **s1** from **s2**, the resulting signal **s12** has minimal energy. The phase of signal **s1**, which also contains omni portion (W), is used in order to provide the subtracted signal with this phase. This process serves to show the original character of the useful signal. A common feature of FIGS. **14** and **15** and FIG. **17** is that an attempt is made to generate a signal **s1** that has an omni portion W, in addition to Figure eight portions X and Y, and the purest possible Figure-of-eight signal **s2**.

The synthesized output signals **s12**(n) contain phase information from the special directions that point to the useful sound source, or are bundled on it; **s1**, whose phase is used, is the signal that has increasing useful signal portions, in contrast to **s2**. Through this analysis, the useful signal is not distorted and retains its original sound.

FIG. **15** shows the synthesized directional characteristics of the individual combined signals M**1**, M**2**, M**3** and the intermediate signals, in which the amplitudes are in each case normalized to the useful sound direction designated with about 0°, (e.g., all the polar curves and those during sound exposure from a 0° direction are normalized to 0 dB). The output signal **1402** then has a directional characteristic bundled particularly strongly in one direction.

Other alternate systems and methods may include combinations of some or all of the structure and functions described above or shown in one or more or each of the figures. These systems or methods are formed from any combination of structure and function described or illustrated within the figures. Some alternative systems or devices compliant with one or more of the transceiver protocols may communicate with one or more in-vehicle or out of vehicle receivers, devices or displays.

The methods and descriptions described above including those shown in FIGS. **1** and **11-18** may be programmed in one or more controllers, devices, processors (e.g., signal processors). The processors may comprise one or more central processing units that supervise the sequence of micro-operations that execute the instruction code and data coming from memory (e.g., computer memory) that generate, support, and/or complete a compression or signal modifications. The dedicated applications may support and define the functions of the special purpose processor or general purpose processor that is customized by instruction code (and in some applications may be resident to vehicles). In some systems, a front-end processor may perform the complementary tasks of gathering data for a processor or program to work with, and for making the data and results available to other processors, controllers, or devices.

The methods and descriptions may also be programmed between one or more signal processors or may be encoded in a signal bearing storage medium a computer-readable medium, or may comprise logic stored in a memory that may be accessible through an interface and is executable by one or more processors. Some signal-bearing storage medium or computer-readable medium comprise a memory that is unitary or separate from a device, programmed within a device, such as one or more integrated circuits, or retained in memory and/or processed by a controller or a computer. If the descriptions or methods are performed by software, the software or logic may reside in a memory resident to or interfaced to one or more processors or controllers that may support a tangible or visual communication interface, wireless communication interface, or a wireless system.

The memory may include an ordered listing of executable instructions for implementing logical functions. A logical function may be implemented through digital circuitry, through source code, or through analog circuitry. The software may be embodied in any computer-readable medium or signal-bearing medium, for use by, or in connection with, an instruction executable system, apparatus, and device, resident to system that may maintain persistent or non-persistent connections. Such a system may include a computer-based system, a processor-containing system, or another system that includes an input and output interface that may communicate with a publicly accessible distributed network through a wireless or tangible communication bus through a public and/or proprietary protocol.

A "computer-readable storage medium," "machine-readable medium," "propagated-signal" medium, and/or "signal-bearing medium" may comprise any medium that contains, stores, communicates, propagates, or transports software or data for use by or in connection with an instruction executable system, apparatus, or device. The machine-readable medium may selectively be, but not limited to, an electronic, magnetic, optical, electromagnetic, infrared, or semiconductor system, apparatus, device, or propagation medium. A non-exhaustive list of examples of a machine-readable medium would include: an electrical connection having one or more wires, a portable magnetic or optical disk, a volatile memory, such as a Random Access Memory (RAM), a Read-Only Memory (ROM), an Erasable Programmable Read-Only Memory (EPROM or Flash memory), or an optical fiber. A machine-readable medium may also include a tangible medium upon which software is printed, as the software may be electronically stored as an image or in another format (e.g., through an optical scan), then compiled, and/or interpreted or otherwise processed. The processed medium may then be stored in a computer and/or machine memory.

While various embodiments of the invention have been described, it will be apparent to those of ordinary skill in the art that many more embodiments and implementations are possible within the scope of the invention. Accordingly, the invention is not to be restricted except in light of the attached claims and their equivalents.

I claim:

1. A microphone arrangement comprising:

a plurality of pressure gradient transducers each having an acoustic center, a first sound inlet opening leading to a front of a diaphragm, and a second sound inlet opening leading to a back of the diaphragm;

a directional characteristic of each of the plurality of pressure gradient transducers comprising an omni portion and a figure-eight portion, the plurality of pressure gradient transducers each having a direction of maximum sensitivity in a main direction, each main direction of the plurality of pressure gradient transducers are inclined relative to each other; and

a pressure transducer having a diaphragm and an acoustic center, the acoustic center lying within an imaginary sphere with each of the acoustic centers of the plurality of pressure gradient transducers, the imaginary sphere having a radius corresponding to double the largest dimension of the diaphragm of the pressure transducer or the plurality of pressure gradient transducers.

2. The microphone arrangement of claim **1** where each of the acoustic centers of the pressure gradient transducers and the pressure transducer lay within an imaginary sphere whose radius corresponds to the largest dimension of the diaphragms of the plurality of pressure gradient transducers.

3. The microphone arrangement of claim **1** where the plurality of the pressure gradient transducers and the pressure transducer are arranged within a boundary.

4. The microphone arrangement of claim **1** where projections of the main directions of the plurality of pressure gradient transducers that lie in a base plane that is spanned by the first sound inlet openings of the plurality of pressure gradient transducers and a sound inlet opening of the pressure transducer enclose an angle between 30° and 150°.

5. The microphone arrangement of claim **4** where the projections of the main directions of the plurality of pressure gradient transducers that lie in the base plane that is spanned by the first sound inlet openings of the plurality of pressure gradient transducers and the sound inlet opening of the pressure transducer enclose an angle of about 90° with each other.

6. The microphone arrangement of claim **1** where for each of the plurality of pressure gradient transducers, the first sound inlet opening and the second sound inlet opening are arranged on the same side of a transducer housing.

7. The microphone arrangement of claim **3** where a front of each of the plurality of pressure gradient transducers and a front of the pressure transducer are arranged flush with the boundary.

8. The microphone arrangement of claim **7** where for each of the plurality of pressure gradient transducers the first sound inlet opening is arranged on a front of a transducer housing and the second sound inlet opening is arranged on a back of the transducer housing.

9. The microphone arrangement of claim **1** where the plurality of pressure gradient transducers and the pressure transducer are arranged within a common capsule housing.

10. A microphone arrangement comprising:

a plurality of pressure gradient transducers each having an acoustic center, a first sound inlet opening leading to a front of a diaphragm, and a second sound inlet opening leading to a back of the diaphragm;

a directional characteristic of each of the plurality of pressure gradient transducers comprising an omni portion and a figure-eight portion, the plurality of pressure gradient transducers having a direction of maximum sensitivity in a main direction, each main direction of the plurality of pressure gradient transducers are inclined relative to each other; and

a pressure transducer having a diaphragm and an acoustic center, the acoustic center lying within an imaginary sphere with each of the acoustic centers of the plurality of pressure gradient transducers, the sphere having a radius corresponding to double of the largest dimension of the diaphragm of the pressure transducer or the plurality of pressure gradient transducers;

where the first sound inlet openings of the plurality of pressure gradient transducers and a sound inlet opening of the pressure transducer lie in a plane, and the second sound inlet openings of the plurality of pressure gradient transducers lie outside the plane, and

where the plurality of pressure gradient transducers consists of two pressure gradient transducers.

11. The microphone arrangement of claim **10** where for each of the plurality of pressure gradient transducers, the first sound inlet opening and the second sound inlet opening are arranged on the same side of a transducer housing.

12. A microphone arrangement comprising:

a plurality of pressure gradient transducers, arranged on an imaginary omni surface, each having an acoustic center, a first sound inlet opening leading to a front of a diaphragm, and a second sound inlet opening leading to a back of the diaphragm;

a directional characteristic of each of the plurality of pressure gradient transducers comprising an omni portion and a figure-eight portion, the plurality of pressure gradient transducers each having a direction of maximum sensitivity in a main direction, each main direction of the plurality of pressure gradient transducers are inclined relative to each other; and

a pressure transducer, arranged on the imaginary omni surface, having a single sound inlet, a diaphragm and an acoustic center, the acoustic center lying within an imaginary sphere with each of the acoustic centers of the plurality of pressure gradient transducers, the imaginary sphere having a radius corresponding to double the largest dimension of the diaphragm of the pressure transducer or the plurality of pressure gradient transducers,

where the first sound inlet opening of the plurality of pressure gradient transducers and the single sound inlet opening of the pressure transducer face each other.

13. The microphone arrangement of claim **12** where the first sound inlet opening of each of the plurality of pressure gradient transducers is arranged on a front of a transducer housing, and the second sound inlet opening of each of the plurality of pressure gradient transducer is arranged on a back of the transducer housing.

14. The microphone arrangement of claim **12** where projections of the main directions of the plurality of pressure gradient transducers that lie in a base plane that is spanned by the first sound inlet openings of the plurality of pressure gradient transducers and a sound inlet opening of the pressure transducer enclose an angle between 30° and 150°.

15. The microphone arrangement of claim **12** where the plurality of pressure gradient transducers are arranged within a boundary.

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